SpoTel: SpoATA series FXO/FXS Voice Gateway

User Manual

Nicherons

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1. Equipment Introduction

1.1 Overview

SpoATA FXO or FXS analog phone adapter (ATA)/gateway provides low cost, simple operation VoIP solutions for enterprise, the family office (SOHO), remote office and branch enterprise. ATA connects to analog telephone, fax and traditional analog PBX with standard voice interfaces and provided high quality voice service. SPOSpoATA series VoIP access gateway adopts standard SIP protocol and is compatible with leading IP PBX (such as Asterisk, 3CS), soft-switch and SIP-based platform. SPOSpoATA series analog gateway includes following model:

- SpoATA-8S (with 8x FXS ports)
- SpoATA-8O (with 8x FXO ports)
- SpoATA-4O (with 8x FXO ports)

This manual is using SptATA-8O as an example, introducing the function of devices and parameter configuration.

1.2 Equipment appearance (back view)

1.3 Power supply

SPOATA-4/8O is desktop equipment and powered by an 12DC power adapter which works between AC 110-240 V

Power adapter specs:

Input: 100-240V, 50-60Hz

Output: 12VDC

1.4Network Applications

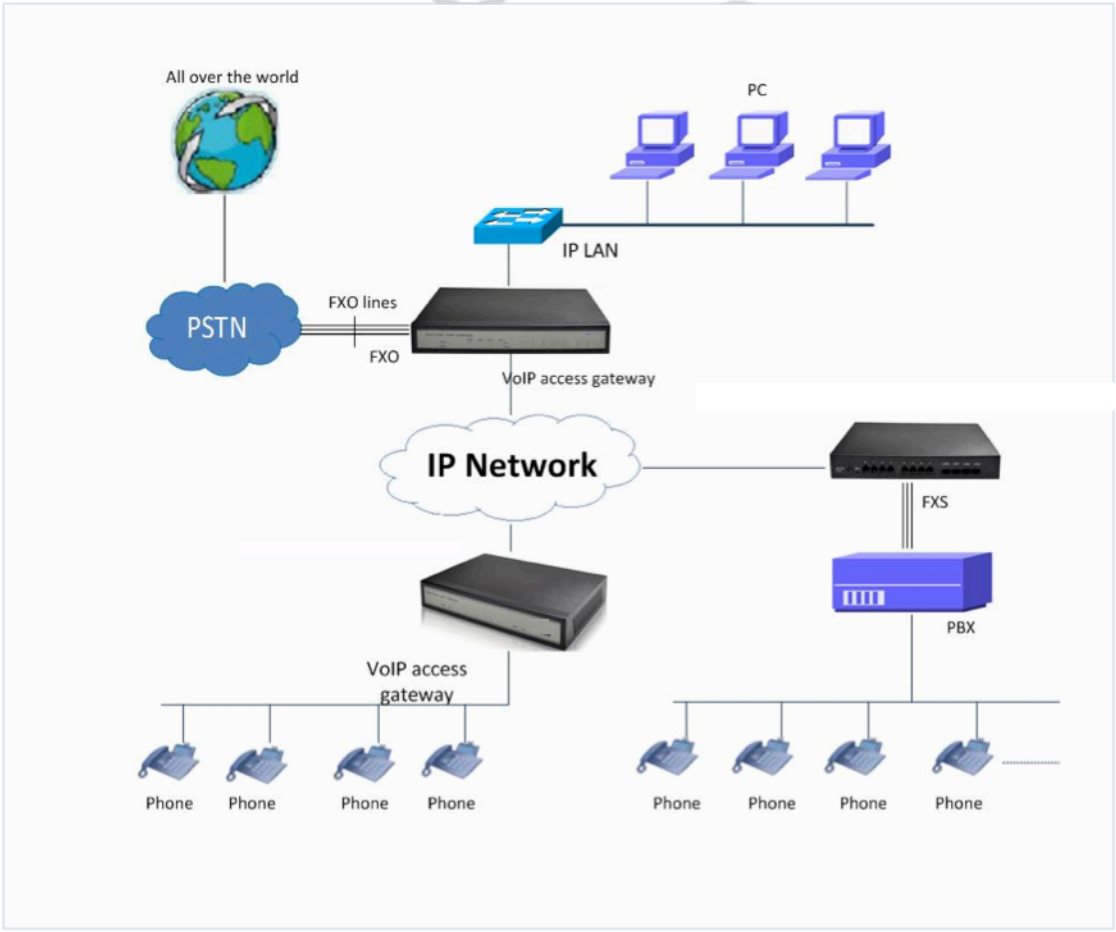

Figure 4-1：Network Applications

1.5 Functions and Features

1.5.1Protocol standard supported

- SIP V2.0 (RFC 3261,3262,3264)
- SDP (RFC 2327)
- REFER (RFC 3515)
- RTP/RTCP (RFC 1889,1890)
- STUN (RFC 3489)
- ARP/RARP (RFC 826/903)
- SNTP (RFC 2030)
- DHCP/PPPoE
- TFTP/HTTP/HTTPS
- DNS/DNS SRV (RFC 1706/RFC 2782)
- VLAN 802.1P/802.1Q
- Diff Server

1.5.2 Voice and Fax parameters

- G.711A/U law, G.723.1, G.729AB
- Comfortable Noise Generation (CNG)
- Voice Activity Detection (VAD)
- Echo Cancellation (G.168)
- Adaptive Dynamic Jitter Buffer
- Voice and fax gain control
- Hook flash
- Modem
- T.38/Pass-through
- DTMF Mode: Signal/RFC2833/INBAND

1.5.3 Supplementary service

- Busy tone detection
- No current take out stitches detection
- Voice interrupted detection
- One stage dialing
- Two stage dialing
- PSTN exterior ports polling
- Polarity Reversal
- FAS（ Fake billing correction ）
- DC/AC impedance config
- Calls detection (Bellcore Type 1&2, ETSI,DTMF)

- Voice mail
- Direct IP Call
- IP Trunk

2. Basic Operations

2.1 Phone Call

2.1.1 Phone or Extension Number

1）FXO Call Out

- One stage dialing: After receiving dialing number from softswitch/IPPBX, the number will be called out via FXO ports that is configured under "Port select".
- Two stage dialing: the call arrives the gateway from SIP connection: the caller will hear 2nd dial tone and could then dial outside number.

2）Dial the number directly and press #.

- Dial outside number with FXO: when the message "please dial the extension number" or second dial tone is played, enter the dialing number. The gateway will then send the number to SIP server/connection, such as soft switch or IPPBX.
- Off-hook auto-dial: (calling via FXO): the gateway will automatically connect to the preset extension or ring group number according to the default hotline number.

2.1.2 Direct IP Calls (SPOATA-8S FXS gateway)

SpoATA-8S allows direct call through IP address. A point-to-point call could be dialed without registration.

Elements necessary to configure a direct IP call：

1) Both SpoATA-8S or other ATA, are on public IP addresses;
2) Both SpoATA-8S and other ATA are on the same LAN on private IP addresses;
3) Both SpoATA-8S and other ATA could be connected through a router using public or

private IP addresses (with necessary port forwarding or DMZ).

Operation Process

1）Pick up the analog phone then dial " *47 "

2）Enter the target IP address.

【Note】: No dial tone will be played between step 1 and step 2

Examples:

If the target IP address is 192.168.0.160, the dialing convention is ***47,** then **192*168*0*160.** Followed by pressing the "#" key or waiting another 3 seconds, the remote unit will ring.

【Note】: Th direct IP calls could not be made from one port to another port within the same gateway since both ports are using the same IP. Only the standard SIP port 5060 could be used in the direct ip dialing.

2.2 Call Features

SpoATA-8S supports all traditional and advanced phone functions.

Table 2.2-1 Feature Codec

Feature Codec	Operation Instructions
***158#**	Inquire the LAN port IP address
***159#**	Inquire the WAN port IP address
***114#**	Inquire port account
150	Set how IP address would be obtained
157	Set network mode
152	Set IP address
153	Set Subnet mask
156	Set default gateway IP address
***193#**	Obtain IP address through DHCP
***160*1#**	Open WAN port to access web
***166*000000#**	Factory reset

***111#**	Restart device
***#**	Call hold
47	IP address call
***51#**	Enable call waiting
***50#**	Disable call waiting
87	Blind transfer
72	Enable Unconditional Call Forward
***73#**	Disable Unconditional Call Forward
90	Enable Busy Call Forward
***91#**	Disable Busy Call Forward
92	Enable No Answer Call Forward
***93#**	Disable No Answer Call Forward
***78#**	Enable DND
***79#**	Disable DND
***200#**	Access Voice mail
Flash/Hook	Switch between incoming calls, If not in session, flash/hook will switch a new channel for new call.

2.3 Sending and Receiving Fax

2.3.1 SpoATA-8S supports four fax modes:

1）T.38（FoIP）

2）Pass-Through

3）Modem

4）adaptive

2.3.2 T. 38 and Pass-Through

T.38 is the preferred method because it is more reliable and works well in most network

conditions. If the service provider supports T.38, please use this method by selecting T.38 as fax mode (default). If the service provider does not support T.38, pass-through mode could be used. If there is problem sending or receiving Fax, toggle the Fax Tone Detection Mode setting.

3. Local IVR Operation

3.1 Inquire IP address

After attaching an to a FXS ports, feature code could be dialed to inspect the current ip status.

: dialing *158# to inquire LAN IP address

: dialing *159# to inquire WAN IP address.

3.2 Factory Reset

Dial *166*000000#, then hang up. The gateway will then reboot.

3.3 Configure LAN Port's IP Address

Before configuration, please ensure:（1）The device is power on; （2）device is connected to the network;（3）Telephone is attached to FXS port.

1) Configure dynamic IP address by DHCP：

Off-hook; Dial "*150*2#"; On-hook/hang up;

After 10 seconds, the device will reboot.

2) Configure Static IP address

Off-hook; Dial "*150*1#"; On-hook;

Then configure IP and mask as follow:

- Configure IP address：

Off-hook; input "*152*172*16*0*100# "; On-hook

- Configure subnet mask：

Off-hook; input "*153*255*255*0*0# "; on-hook

- Configure gateway IP address

Off-hook; input "*156*172*16*0*1# "; on-hook.

3) Query the IP address of device: Off-hook, input"*158#"
4) If the SpoATA-8S is on PPPoE to get IP address，it should be configured by web browser.

【Note】: The telephone will play voice prompt "Setting successfully" if the setup is correct

4. WEB Configuration (all SpoATA series)

4.1 WEB Login

First to connect the Device to network properly, refer to chapter 3 "Operation". Off-hook and dial*158# to inquire its IP address.

4.1.1 Login

The LAN port default IP address is 192.168.11.1, WAN port will obtain IP address by DHCP. First modify your pbx to the same IP segment/domain, with Windows 7 as an example, the computer IP could be set as 192.168.11.10：

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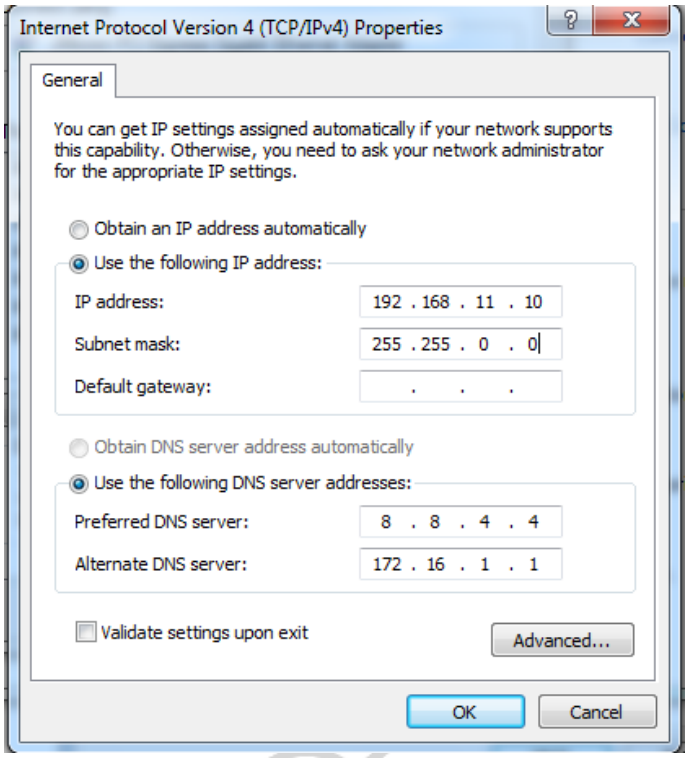
Internet Protocol Version 4 (TCP/IPv4) Properties
General
You can get IP settings assigned automatically if your network supports this capability. Otherwise, you need to ask your network administrator for the appropriate IP settings.
Obtain an IP address automatically
Use the following IP address:
IP address: 192 . 168 . 11 . 10
Subnet mask: 255 . 255 . 0 . 0
Default gateway:
Obtain DNS server address automatically
Use the following DNS server addresses:
Preferred DNS server: 8 . 8 . 4 . 4
Alternate DNS server: 172 . 16 . 1 . 1
Validate settings upon exit
Advanced...
OK
Cancel

Figure 4.1-1Modify IP address

Check connection between computer and device, click "Start"-> "run"-> input "cmd", run " ping 192.168.11.10 –t " to verify the connectivity between them.

4.1.2 Login WEB

Open web browser, then enter IP address The login interface will appear.

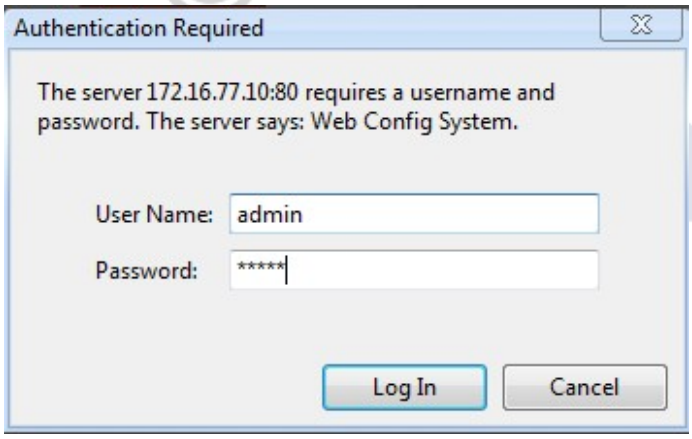
Authentication Required
The server 172.16.77.10:80 requires a username and password. The server says: Web Config System.
User Name: admin
Password: *****
Log In
Cancel

Figure 4.1-1 Login Interface

Default username and password: admin/admin, click "OK" to enter into web management interface.

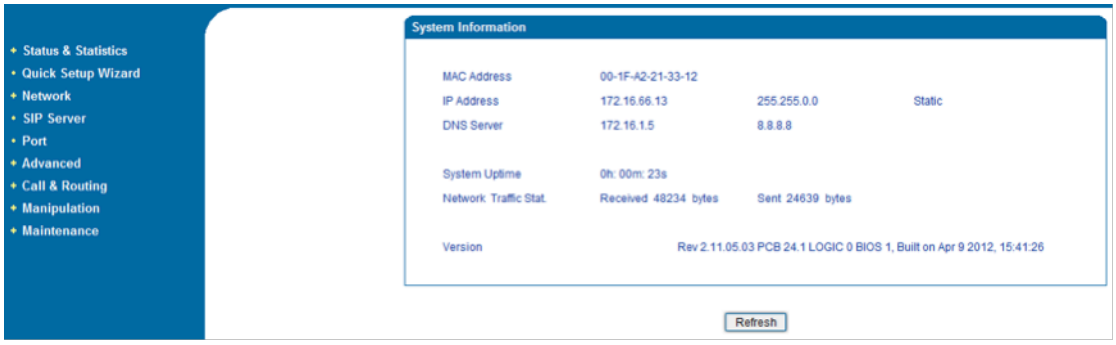

Figure 4.1-2 Configuration Interface

4.2 Configuration menu

SpoATA web management interface displays in the left. Navigate the menu to complete the configuration.

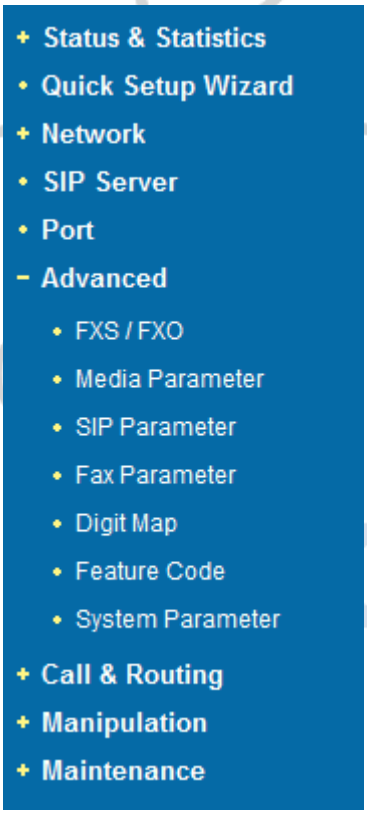

Figure 4.2-1 Navigation Tree

When LAN port is set at bridge mode, the several sub-menus will not display, including "Routing configuration", "DHCP service", "DMZ host", "forward rules" and "static routing" and "ARP" etc.

4.3 Status and Statistics

4.3.1 System Information

System information interface shows the run information as following figure 4.3.1 below:

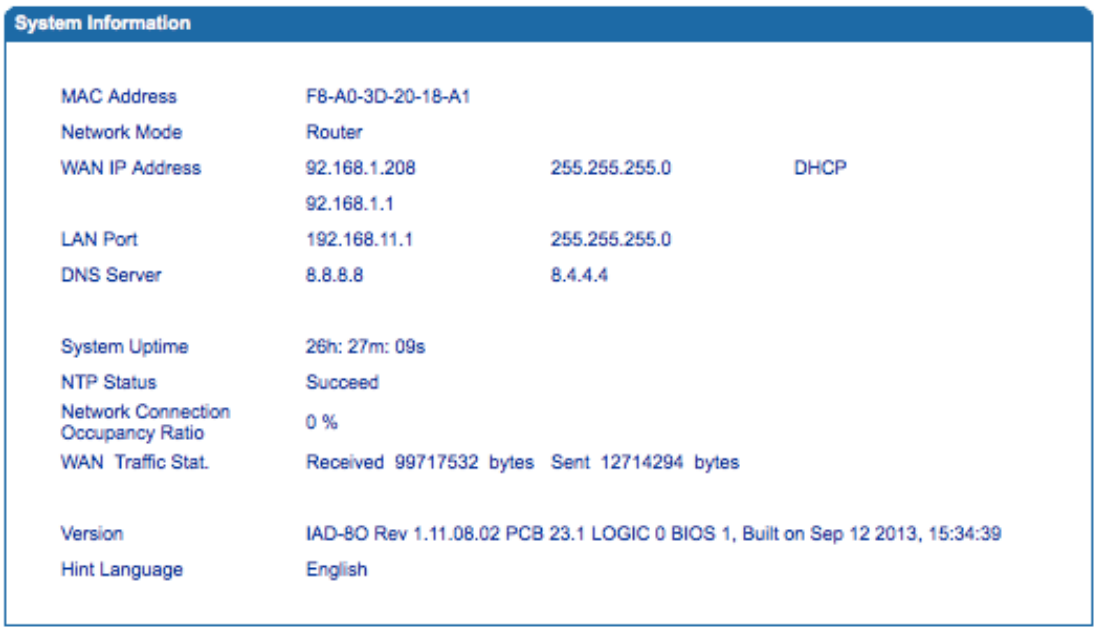

Figure 4.3-1 System Information

System information as follow:

Table 4.3-1 System Information Description

MAC address	WAN port hardware address. The device ID in HEX format.
Network Mode	Display network mode, include bridge and rout. If it is bridge, WAN port display Network, and the WAN port IP as same as the LAN port IP.
WAN Port	Shows WAN IP address of ATA , DHCP mode: all the field values for the Static IP mode are not used (even though they are still saved in the Flash memory.) The ATA acquires its IP address from the first DHCP server it discovers from the LAN it is connected. Using the PPPoE feature: set the PPPoE account settings. The ATA will establish a PPPoE session if any of the PPPoE fields is set. Static IP mode: configure the IP address, Subnet Mask, Default Router IP address, DNS Server 1 (primary), DNS Server 2 (secondary) fields. These fields are set to zero by default.
LAN Port	Shows LAN IP address. If network Mode is bridge, LAN port won't display.
DNS Server	Display DNS server IP address and default gateway information
System Uptime	Time elapsed from device power on to now.

Network Traffic Statics	Total bytes of message received and sent by network port.
Version	Includes: product mode, software version, hardware version and built time etc.

4.3.2 Registration Information

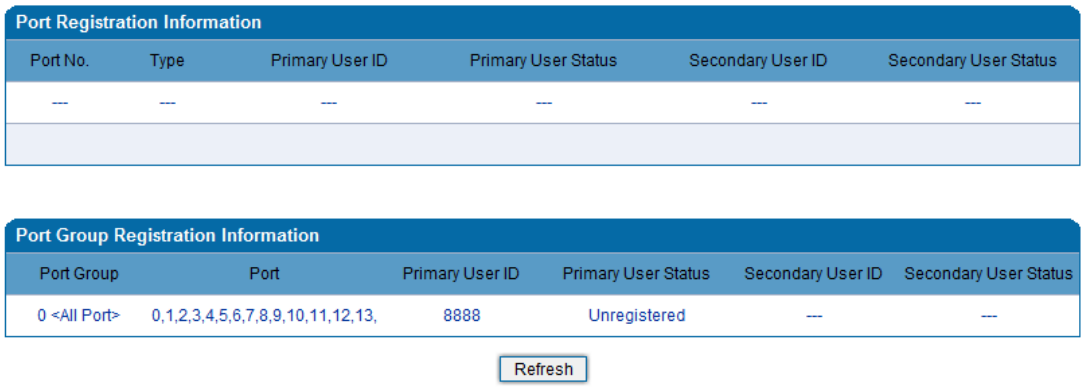

Figure 4.3-2 Port and Port group registration information

4.3.3 TCP/UDP Statistics

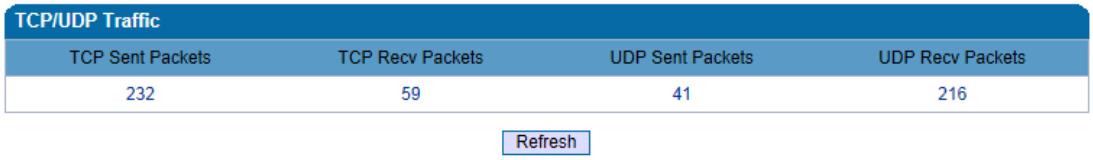

Figure 4.3-3 TCP/UDP Statistics Information

Figure 4.3-3 shows TCP sending and receiving, UDP sending and receiving packets of statistical information since the device launched.

4.3.4 RTP Session Statistics

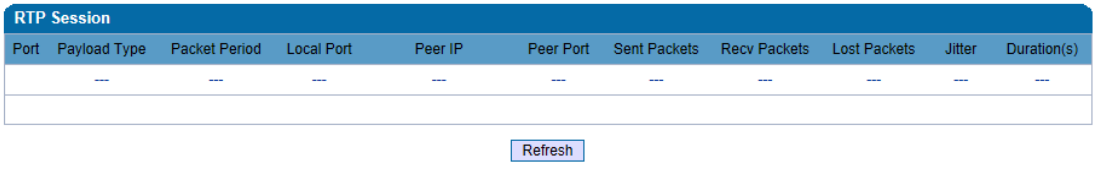

Figure 4.3-4 RTP Session Statistics

Figure 4.3-4 display real-time RTP conversation flow data information, including:

Port, voice codec, packet period, local port, peer IP, peer port, sent packets, receive packets, lost packets, jitter and duration.

4.4 Quick Setup Wizard

Quick configuration wizard will guide users to configure the device step by step. Users only need to configure network, SIP server and sip port during the setup. Basically, after these three steps, users are able to make voice call through the gateway.

4.5 Network Configuration

4.5.1 Local Network

There are two modes, namely, route and bridge. When the gateway is set as route mode, the gateway will work as small router and NAT function is enabled. In this setup, WAN port normally connect to uplink router/switch or ADSL MODEM, while LAN port connects local computer or other network device (such as Ethernet switches, Hubs etc). When bridge mode is set, WAN and LAN ports are daisy chained together as in Ethernet switch. In bridge mode, only WAN IP address and DNS should be configured. In route mode, default LAN IP is 192.168.11.1 but could be changed by users.

Network configure interface as below:

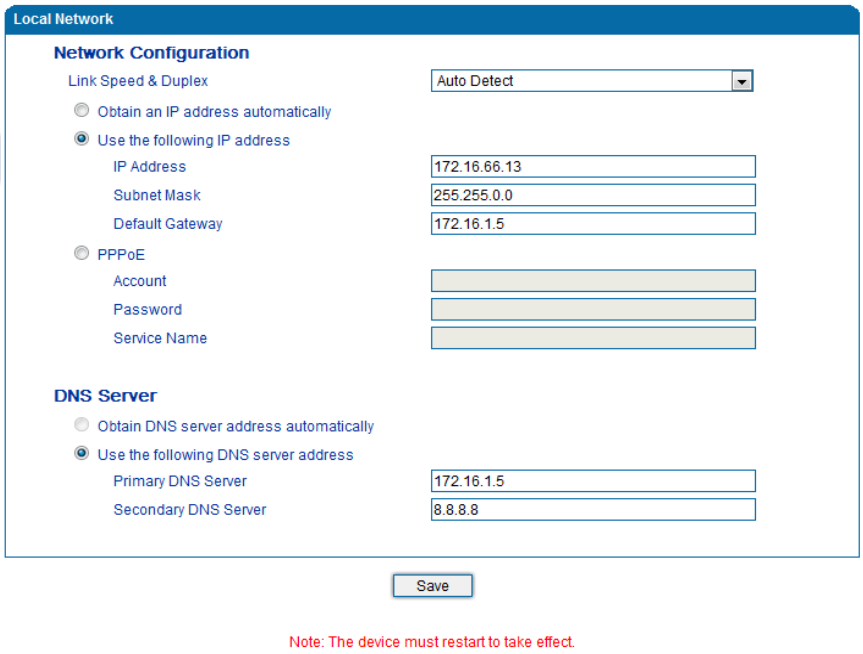

Figure 4.5-1 Local network

- "Link Speed & Duplex" used to select Ethernet mode, include 5 selections between, "Auto Detect" (default)、"10Mbps half-duplex"、"10Mbps full-duplex", "100Mbpshalf-duplex", "100Mbps full-duplex".
- When select" obtain IP address automatically", ATA will obtain IP address by DHCP.
- When select "Use the following IP address", fixed IP address will be set manually.
- When select "PPPoE", please fill in account and password from your .

【Notes】:

1） If DHCP is slected, please verify that working DHCP server is in network.

2） After the network configuration, restart device configuration validation.

4.5.2 VLAN Parameter

Quality of service is particularly important for the transport of traffic with special requirements. In particular, much technology has been developed to allow computer networks to become as useful as telephone networks for audio conversations, as well as supporting new applications with even stricter service demands.

1) 802.1Q

The IEEE 802.1Q standards define architecture , services , protocols and algorithms for Virtual Bridged LANs.

No Quality of Service mechanisms are defined in this standard, but an important requirement for providing QoS in this standard, e.g. ability to regenerate user priority of received frames using priority information contained in the frame and the User Priority Regeneration Table for the reception Port.

2) 802.1p

IEEE 802.1p standards, Traffic class expediting and dynamic multicast filtering:

It describes important methods for providing QoS at MAC level. IEEE 802.1p is very effective. Packets will be sent according to the priority level. IEEE 802.1p describes no

admission control protocols. It would be possible to give Network Control priority to all packets and the network would be easily congested.

There are three VLAN: data VLAN, voice LAN and management VLAN. VLAN configuration interface as following figure 4.5-3:

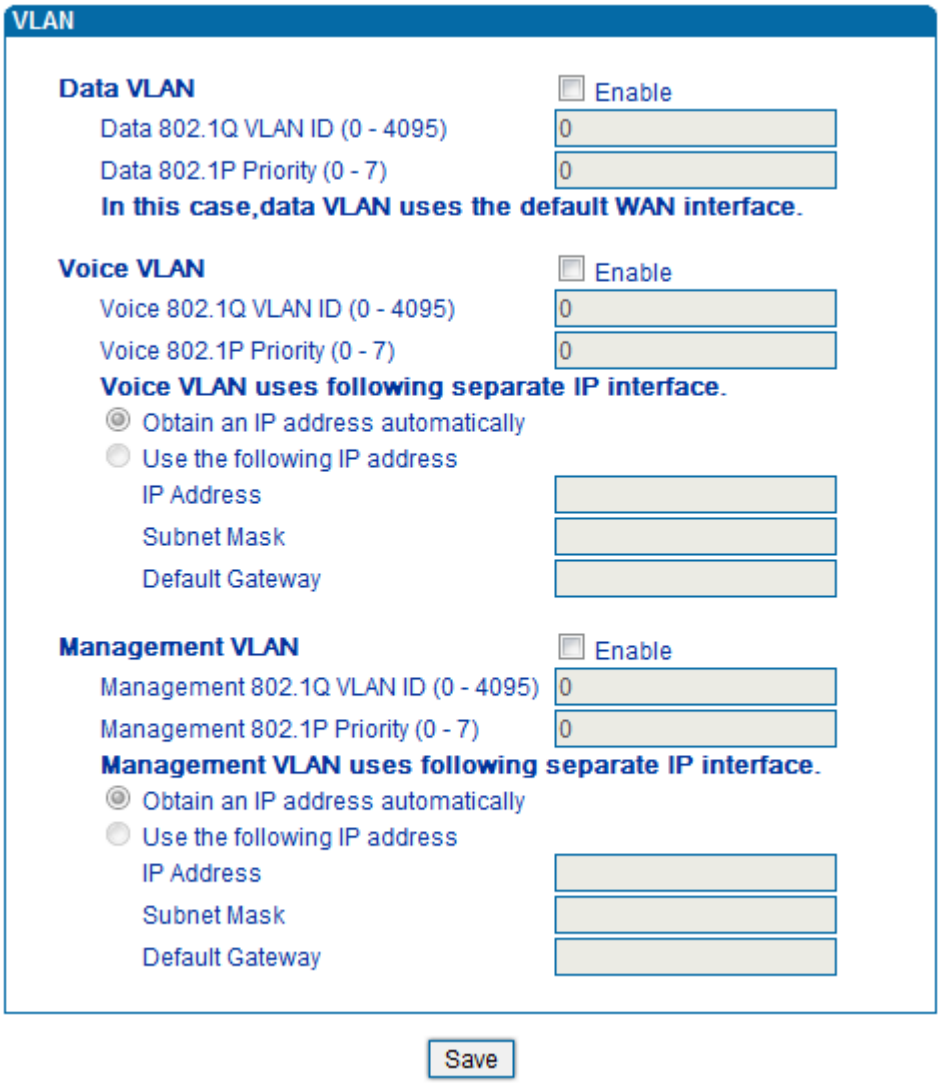

Figure 4.5-3 VLAN parameter configurations

Table 4.5-1VLAN parameter configurations

Data VLAN	Data 802.1Q VLAN ID(0-4095)	Fill out an ID to describe a data VLAN group, ID 0 used to management VLAN, can't used to service configure.
	Data 802.1p Priority（0-7）	802.1 protocol to control network traffic priority, Priority from 0-7.
Voice VALN	Voice 802.1Q VLAN ID(0-4095)	Fill out an ID to describe a voice VLAN group, ID 0 used to management VLAN, can't used to service

		configure.
	Voice 802.1p Priority（0-7）	802.1 protocol to control network traffic priority, Priority from 0-7.
	Voice VLAN use following separate IP interface	Can use dynamic or static IP address
	Voice VLAN DNS Server	Can use dynamic or static DNS server address
Management VLAN	Management 802.1Q VLAN ID(0-4095)	Fill out an ID to describe a data VLAN group, ID 0 used to management VLAN, can't used to service configure.
	Management 802.1p Priority（0-7）	802.1 protocol to control network traffic priority, Priority from 0-7.
	Management VLAN use following separate IP interface	Can use dynamic or static IP address
	Management VLAN DNS server	Can use dynamic or static DNS server address

【Note】: Restart the device to apply changes

4.5.3 ARP

ARP brief introduction:

ARP is address resolution protocol. After configuring ARP, users can get physical address through device IP address. Under TCP/IP network environment, each host is assigned a 32-bit IP address. ARP is a tool that converts IP address into MAC address.

ARP configuration interface as follows:

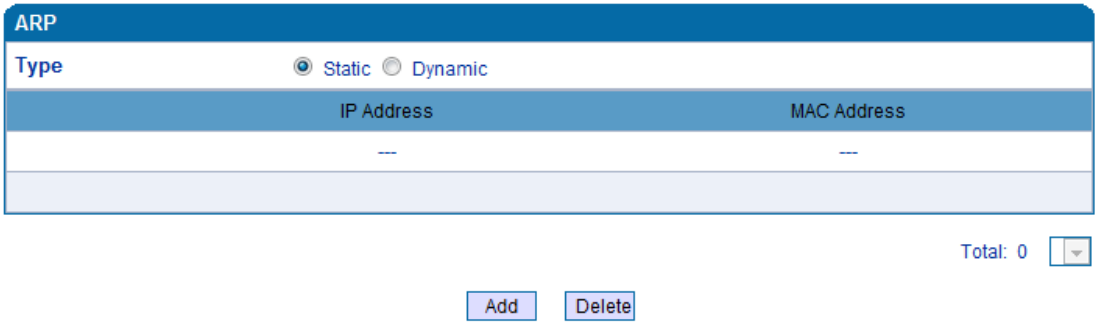

Figure 4.5-4 ARP Parameters

4.6 SIP Server

SIP server introduction:

1）SIP server is the main component of VoIP network and responsible for establishing all the SIP phone calls. SIP server also called SIP proxy server or registered server. IPPBX and the soft-switch can act as SIP server.

2）Usually, SIP server does not participate in the media process.

In SIP network, the media always using end-to-end to hand the consultation. In some particular situation or business processing such as "Music On Old", SIP server will actively participate in the media negotiation. Simple SIP server is responsible only for establishment, maintenance and cleaning conversation, don't interfere in call. While relatively complex SIP server also called SIP PBX. It does not only provide the basic call connection and basic conversational support, but also offer plenty of business features such as: Presence, Find-me, Music On Hold.

3）SIP server based on Linux platform, such as: SpoTel, OpenSER、sipXecx，VoS，Mera etc.

4）SIP server based on windows platform, such as :miniSipServer、Brekeke，VoIPswitch etc.

5）Carrier grade soft-switch platform, such as Cisco, Huawei, Zteetc.

SIP server configuration interface as follows:

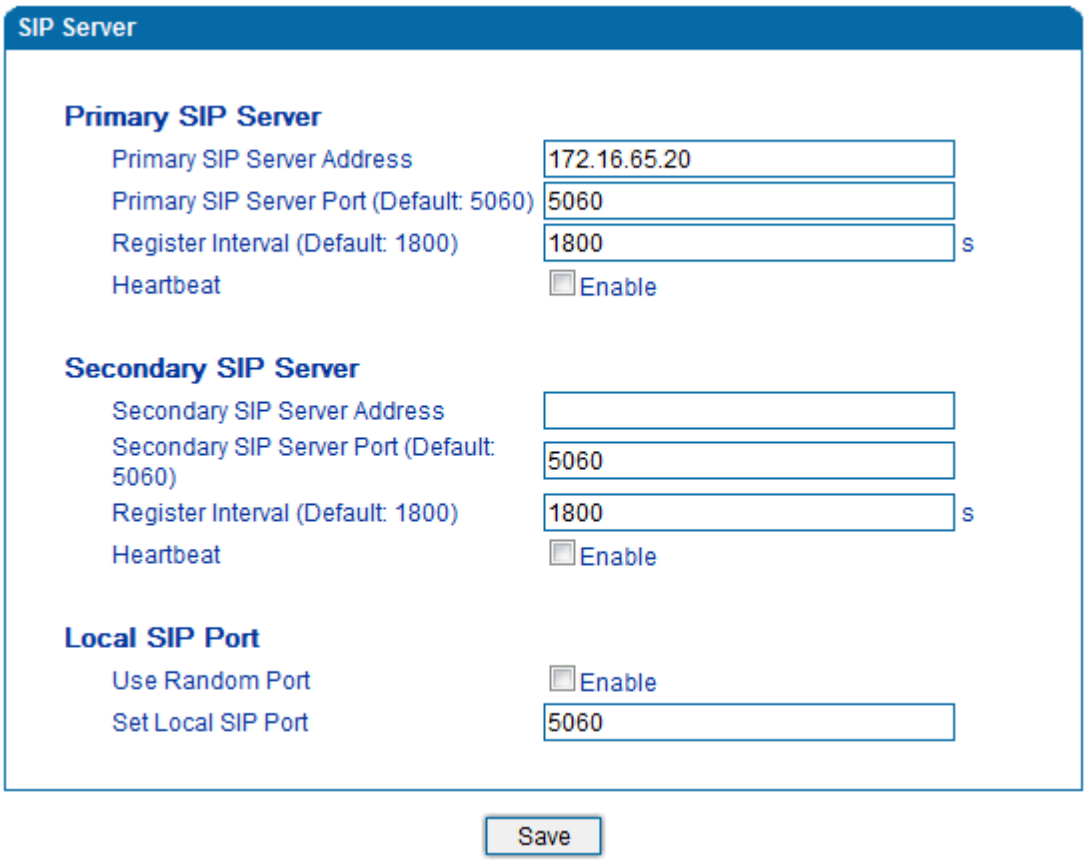

Figure 4.6-1 SIP Server Configuration Interface

SIP parameter description:

Primary SIP Server IP	SIP Server IP address or Domain name provided by VoIP service provider.
Primary SIP Server port	Service port, default is 5060
Register interval	protects registrar against excessively frequent registration refreshes while limiting the state. Every once in a while send request for registration to the terminal server, default is 1800s.
Heartbeat	Heartbeat message detect the connection status between device and SIP server.
Secondary SIP Server IP address	Backup SIP Server's IP address or Domain name provided by VoIP service provider.
Secondary SIP Server port	Service port, default is 5060
Secondary SIP server Register interval	protects registrar against excessively frequent registration refreshes while limiting the state. Every once in a while send request for registration to the terminal server, default is 1800s.
Secondary SIP heartbeat	Heartbeat message detect the connection status between device and SIP server.
Use Random Port	Random SIP service ports for ATA
Set Local SIP port	Default SIP service port is 5060.

4.7 Port Configuration

Port parameters include: Send gain, receive gain, primary display name etc.

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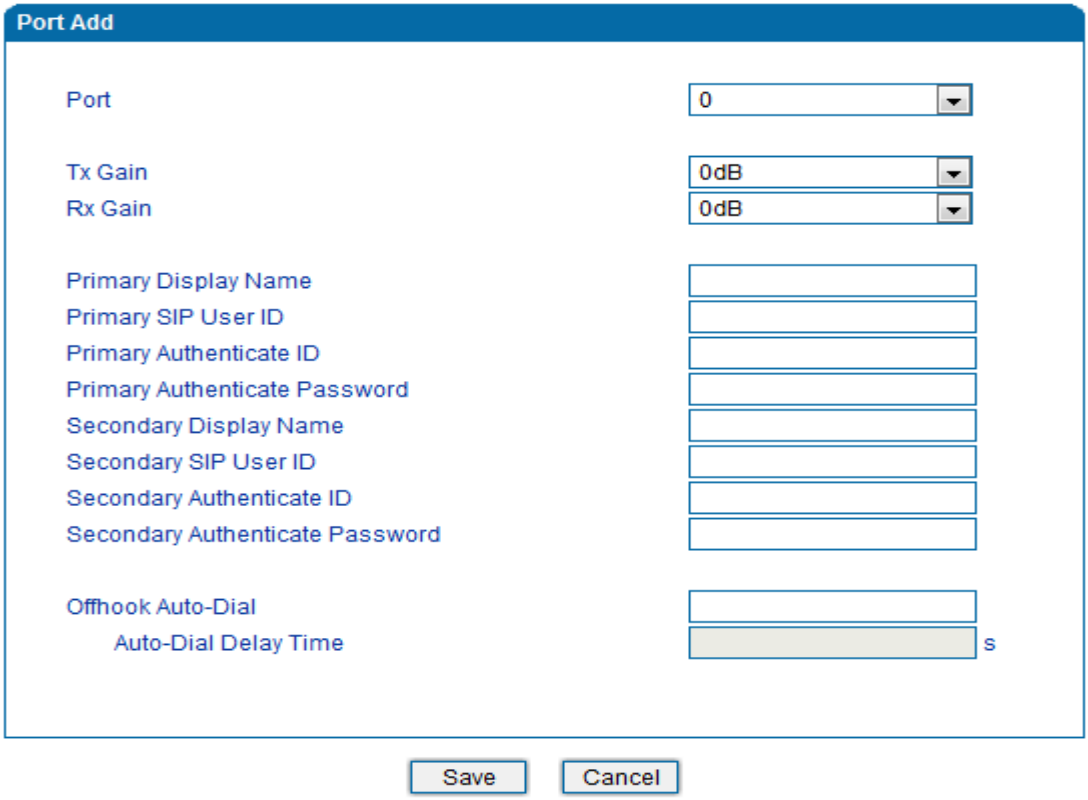


Figure 4.7-1 Port configuration interface

Port parameters introduce as follows:

Tx Gain	It is use to control the volume of conversation, Adjust "TX gain" will affect the end users voice size, the default value is 0. Its value range from-10 – 10 dB
Rx Gain	It is use to control the volume of conversation, Adjust "RX gain" will affect the end users voice size, the default value is 0. Its value range from -10 – 10 dB
Primary /Secondary SIP Display Name	Primary /Secondary SIP account description. Its purpose is so you can identify the SIP account with a meaningful name
Primary /Secondary SIPUser ID	User account information, provided by VoIP service provider (ITSP). Usually in the form of digit similar to phone number or actually a phone number.
Primary/Secondary SIP Authenticate ID	SIP service subscriber's Authenticate ID used for authentication. Can be identical to or different from SIP User ID.
Primary/Secondary Authenticate password	SIP password which registers to soft switch/SIP server
Off-hook Auto-dial	Pre-assign an extension or phone number so that automatically dial a number as soon as you pick up the phone set
Auto-dial Delay Time	Delay 0-3 seconds to automatically dial a number, 0 means dial number immediately

4.8 Advanced

4.8.1 FXO parameters

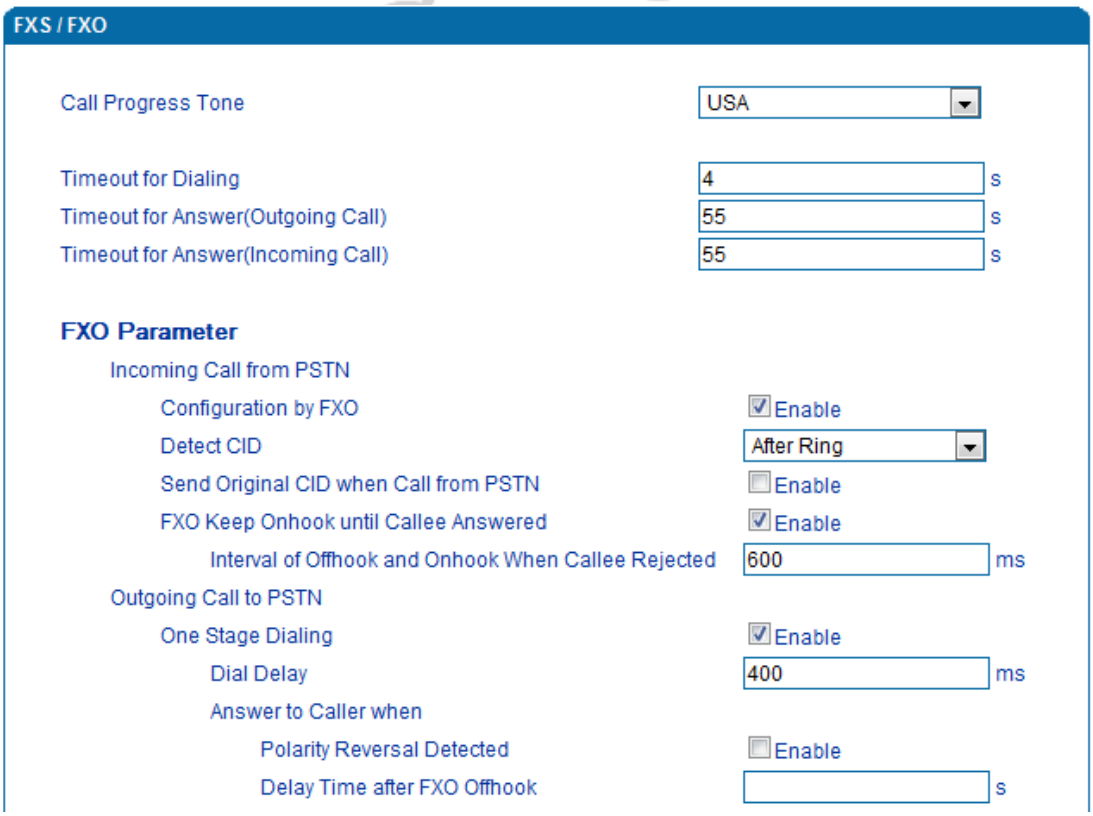
FXS and **FXO** are the name of the ports used by analog phone lines (also known as POTS – Plain Old Telephone Service) or phones.

FXS – **F**oreign e**X**change **S**ubscriber interface is the port that actually delivers the analog line to the subscriber. In other words it is the 'plug on the wall' that delivers a dial tone, battery current and ring voltage.

FXO – **F**oreign e**X**change **O**ffice interface is the port that receives the analog line. It is the plug on the phone or fax machine, or the plug(s) on your analog phone system. It delivers an on-hook/off-hook indication (loop closure). Since the FXO port is attached to a device, such as a fax or phone, the device is often called the 'FXO device'.

FXO and FXS are always paired, i.e similar to a male / female plug.

FXO parameters include: Call progress Tone, Timeout for Dialing, Send Polarity Reversal etc. Configuration interface as follow:



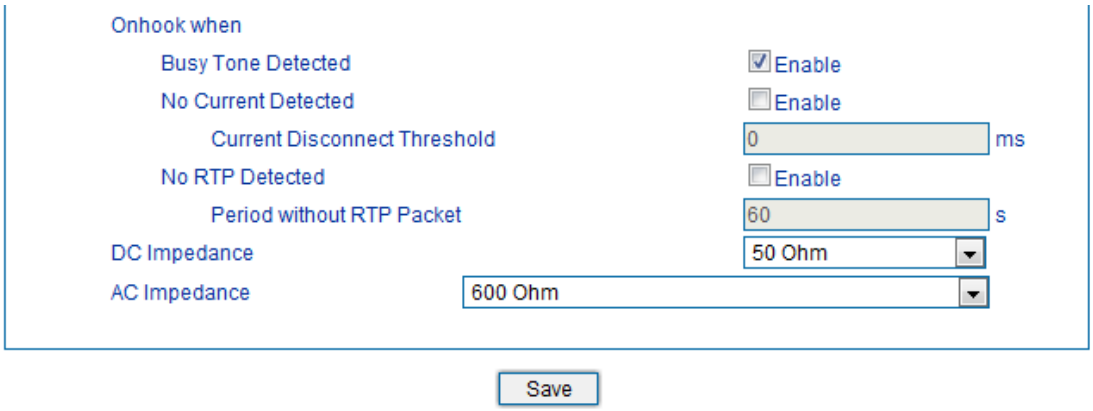


Figure 4.8-1 FXS Parameters Configuration Interface

FXO parameters description:

Call Process Tone	Hear the dial tone when pick up the phone. Choose the national standards from the drop-down box. Default is the United States.
Timeout for dialing	With the help of dialing timeout, you can limit the time while users typing the digits from an extension. If the timeout expire while the user is typing in the extension then ATA will consider the extension as complete and it will try to send to SIP server. Default value is 4 seconds
Timeout for answer(Outgoing call)	This timer set how long the caller party waiting when makes outgoing call on extension.
Timeout for answer(Incoming call)	This timer set how long the phone sets ringing when get incoming call
Configuration by FXO	When the call from FXO interface, users can be enable or disabled FXO allocation function. FXO configuration function includes: detect CID, Send original CID, Play hint to FXO.
Detect CID	FXO detection calling number and the order of the ring. System has two modes: Before ring and After ring.
Send Original CID when Call from PSTN	Enable this function, the extension call display will show the PSTN side number. Otherwise, the call display will show FXO port number.
FXO Keep On-hook until Caller Answered	Enable this function, when call from PSTN to FXO port, FXO port set auto-dial, and pick up after the extension number connection. This function mainly used to billing.
Play Hint to FXO	Enable this function, when call from PSTN to FXO port, FXO port will play prompt tone "please dial the extension number".
One Stage Dialing	Enable this function, FXO port directly sent the dial number, without call extension.
Dial Delay	When call from FXO port to PSTN, the interval of sending number by FXO port, default is 400ms.
Polarity Reversal Detected	When call from FXO port to PSTN, the way of FXO response caller is detecting polarity reversal. If device detected polarity reversal, and then reported to caller to respond, and began to billing.
Delay Time after FXO Off-hook	The time of responding caller by FXO port should be less than this configuration.

Busy Tone Detected	One of the FXO On-hook conditions. When FXO port detected busy tone, FXO will On-hook.
No Current Detected	Another of the FXO On-hook conditions. When FXO port detected no current, FXO will On-hook.
Current Disconnect Threshold	Default the time of no current should be less than 200ms.
No RTP Detected	Enable this function, the system will detect whether RTP flow is interrupted. When voice interrupted, this function can prevent FXO port hanged.
Period without RTP Packet	How long time no RTP packet allowed.
AC/DC Impedance	Adjust impedance, used to impedance matching when FXO and PBX docking.

4.8.2 Media Parameter

Media parameters include: RTP start port, DTMF parameter, Preferred codec. Configuration Interface as follow：

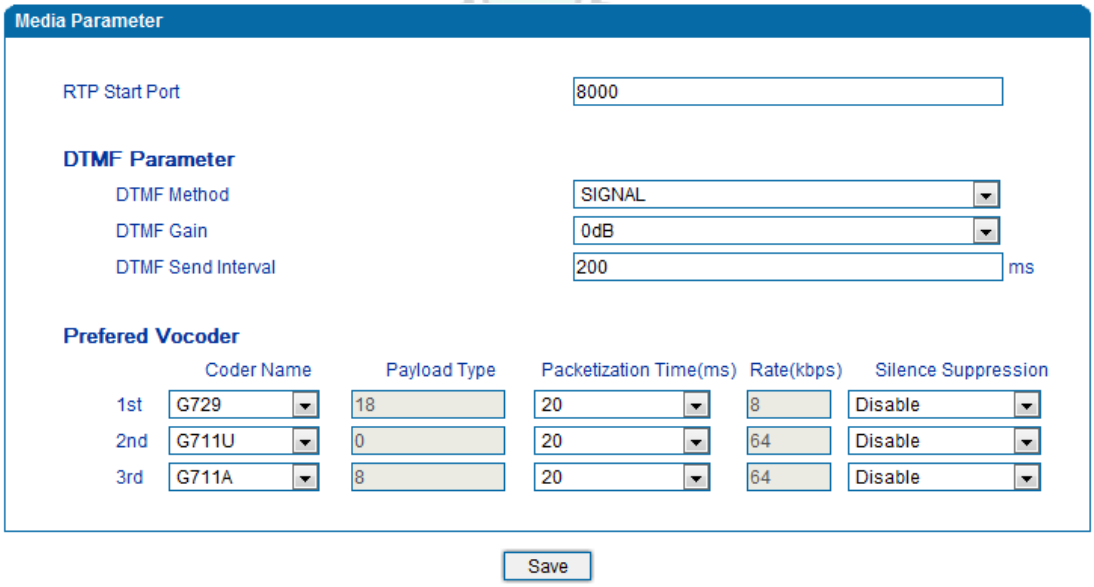


Figure 4.8-2 Media Parameter Configuration Interface

Media parameter description:

RTP Start Port	Default RTP port 8000
DTMF Method	SINGAL、INBAND、RFC2833
RFC2833 Payload Type Optimization	It is configurable When RFC2833 is selected, payload negotiation parameter with remote side, it includes two options: Local and remote
RFC2833 Payload Type	Payloadvalue, default is 101
DTMF Gain	Default is 0 DB
DTMF Send Interval	DTMF send signal interval, default is 200ms.
Coder Name	ATA supports G729、G711U、G711A、G723. while it make outgoing call, G.729 will used as figure 4.8.2 displayed
Payload Type	Each kind of coding has a unique type load value, refer toRFC3551
Packetization Time	Voice package time
Rate	Voice data flow rate, system default
Slience Suppression	Default is disable, if enable, according to the current noise environment dynamically adjust mute inhibit threshold,thus in the user in silent state stop transmission background noise bag and save about VoIP bandwidth.In the low bandwidth environment, can reduce the network congestion, greatly improving VoIP call effect.

Nicherons

4.8.3 SIP Parameter

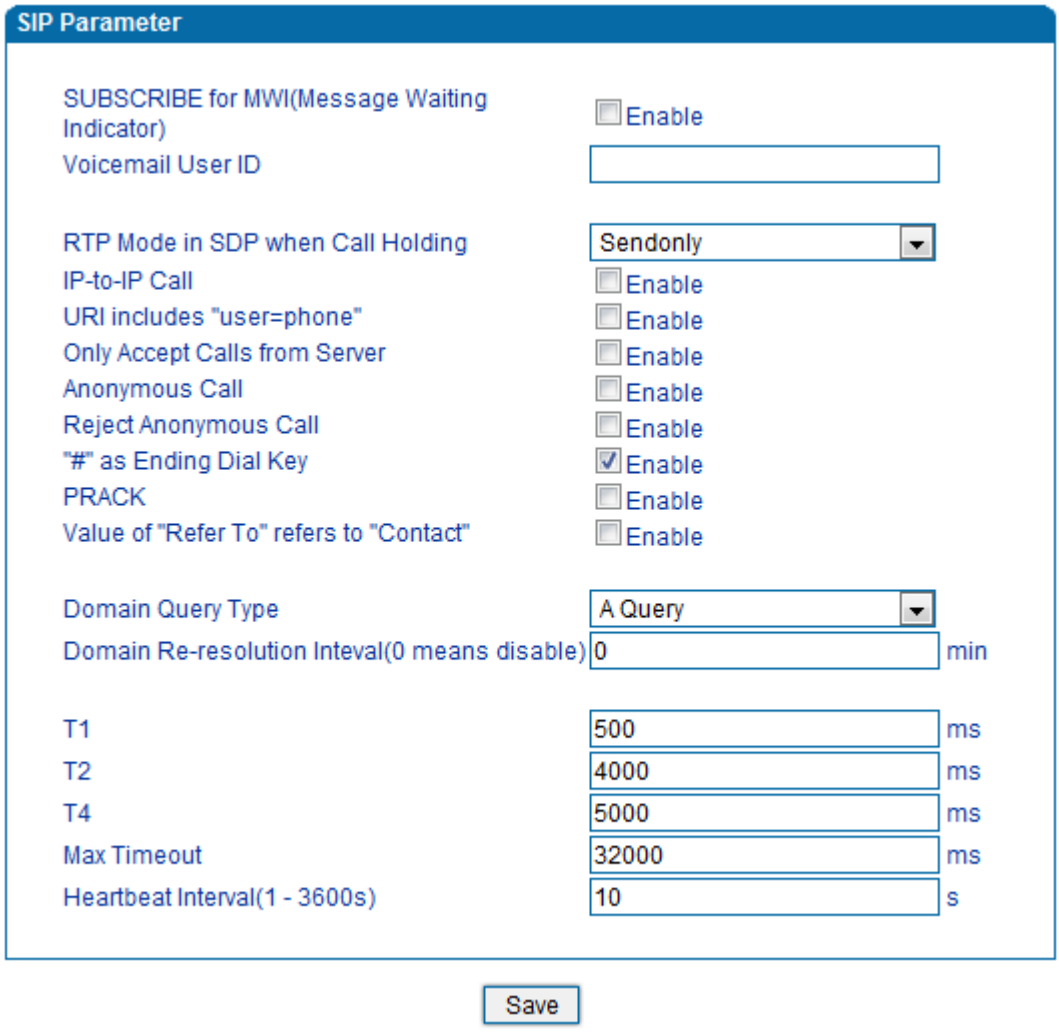


Figure 4.8-3 SIP Parameter Configuration Interface

SIP parameter description：

SUBSCRIBE for MWI	Voicemail message indicator, it is to be realized in the way of NOTIFY
Voicemail User ID	Access code to voicemail box
RTP Mode in SDP when Call Holding	When call come into holding, if select to receive and not send packet, then the local can hear call waiting tone. If select to not receive and not send packet, then doesn't play call waiting tone.
IP-to-IP Call	Enable this function, users may use the * business call IP address on the phone.
URI Includes user=phone	SIP carries the information, the system defaults not open.
Only Accept Call from Server	Default is no, it indicates the ATA accept incoming call from SIP server only

Anonymous Call	Enable anonymous call, "anonymous" will include in SIP message
Reject Anonymous Call	Enable this function, reject all anonymous call. Disable by default
# as ending Dial Key	Dial-up, use # as a end descriptor.
PRACK	RFC3262 defined an optional extension methods—PRACK （provisional ack）, Used to support the reliability of the temporary response.
Value of "Refer To" refers to "Contact"	Its function is to require the receiving party contact with the third party through the use of supplied in the request in the address information. "Refer to" field of SIP message fill in "contact header".
Domain Query Type	There are two modes option: A QUERY and SRV QUERY. Default is A QUERY.
Domain Re-resolution Interval	Default 0: forbidden
T1	T1 timer of SIP protocol, default is 500ms
T2	T2 timer of SIP protocol, default is 400ms
T4	T4 timer of SIP protocol, default is 500ms
Max Timeout	The max timeout of sending or receiving, default is 32s
Heartbeat Interval	Default is 10s.

Nicherons

Voice mail instructions:

Here is to use Elastix to illustrate voicemail works.

1）SpoATA registers to Elastix server. In Elastix, voicemail in enabled & voicemail password is set.

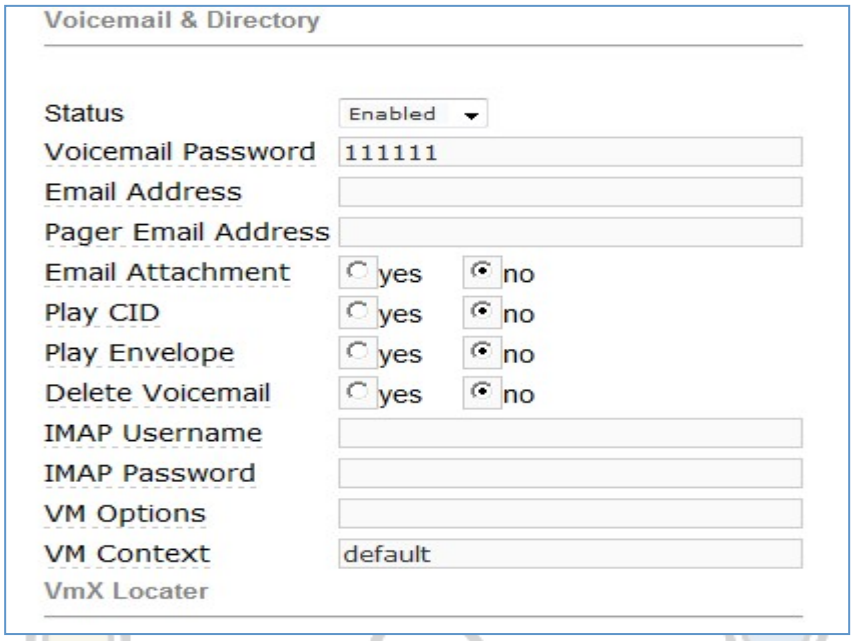


Figure 4.8-4 Elastix Voicemail Configuration Interface

2）check feature code in Elastix and change it as necessary. Its default feature codes are as the below:

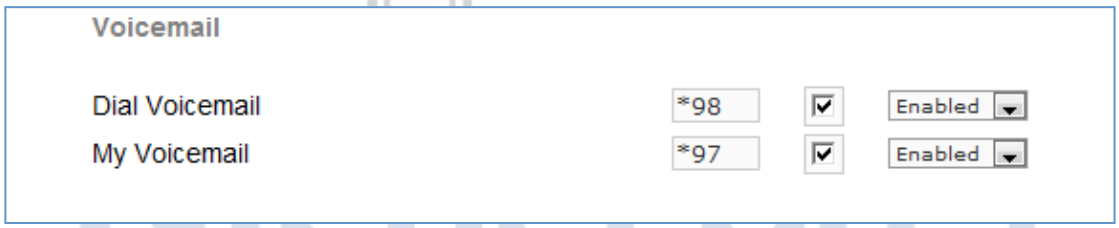


Figure 4.8-5 Elastix Voicemail Setting

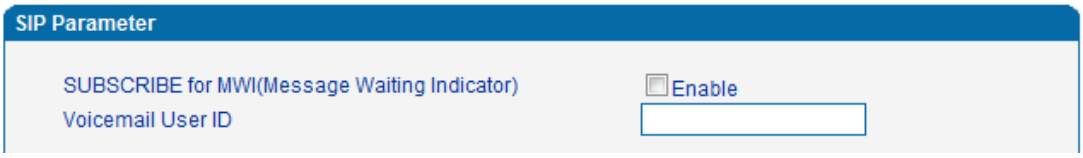


Figure4.8-6 Voice Mail Setting In SIP Parameter

3）Enable voice mail in SpoATA, Elastix will ask you to leave message after 15 seconds, then Elastix will record and display your message (in this example).

Figure 4.8-7 Voicemail Setting

4）Dial *200# from SPoATA to access voicemail. Voicemail number & its password are required before voice message will be played.

4.8.4 Fax Parameter

Fax introduction：

Fax parameters include: fax mode, Fax sound detection party, ECM, Rate.

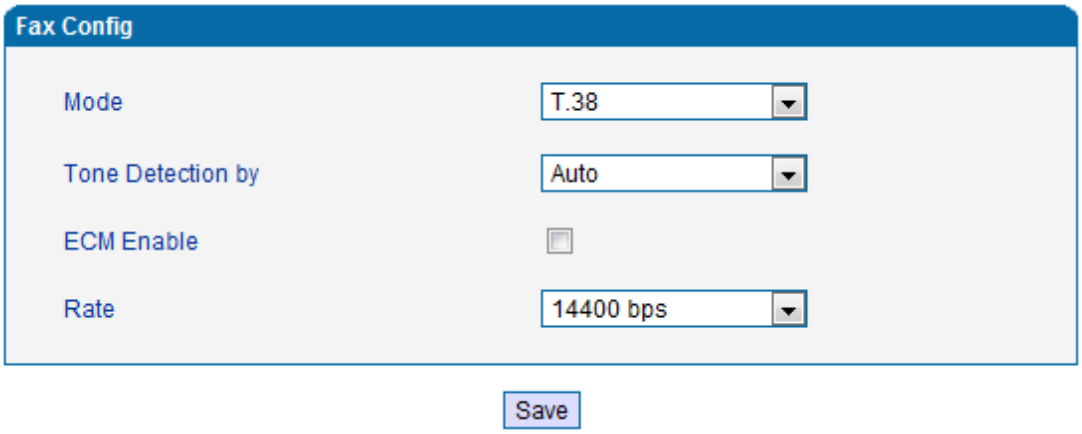


Figure 4.8-8 Fax Parameter Configure Interface

Fax parameter description:

Mode	Fax mode support T.38, T.30 (Pass-through),Modem, Adaptive.
Tone Detection by	Fax sound detection mode: Caller, Caller, Automatic.
ECM	Fax error correction information
Rate	The rate of sending and receiving.

4.8.5 Digit Map

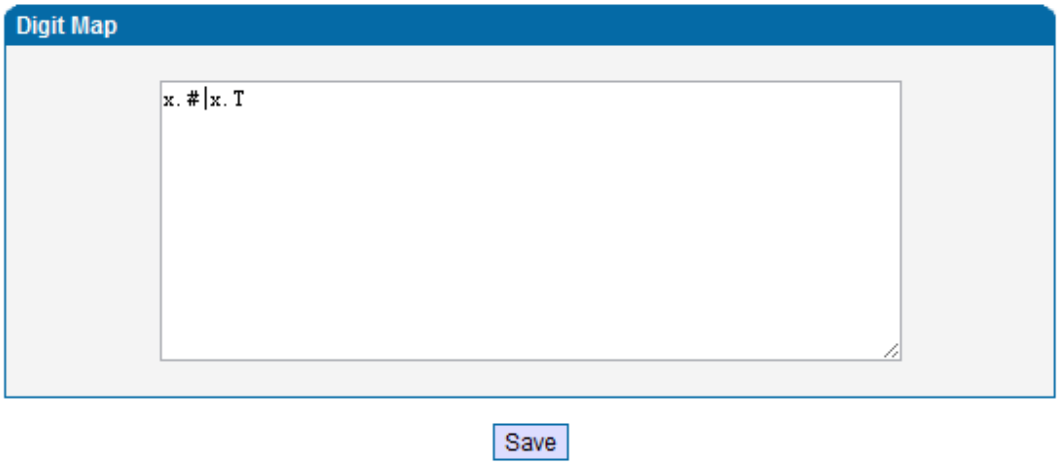


Figure 4.8-9Digit Map

The digit map is the setting that describes different patterns of numbers. When a number matches a pattern, the number is sent to SpoATA to place the call. The pattern may include a timer at the end. If no numbers are entered before the time expires, the number matching the pattern will be sent. If additional numbers are entered before the time elapses, the pattern no longer matches.

Digit Map Syntax:

1. Supported objects

Digit: A digit from "0" to "9".

Timer: The symbol "T" matching a timer expiry.

DTMF: A digit, a timer, or one of the symbols "A", "B", "C", "D", "#", or "*".

2. Range []

One or more DTMF symbols enclosed between square brackets ("[" and "]"), but only one can be selected.

3. Range ()

One or more expressions enclosed between round brackets ("(" and ")"), but only one can be selected.

4. Separator

|: Separated expressions or DTMF symbols.

5. Subrange

-: Two digits separated by hyphen ("-") which matches any digit between and including the two. The subrange construct can only be used inside a range construct, i.e., between "[" and "]".

6. Wildcard

x: matches any digit ("0" to "9").

7. Modifiers

.: Match 0 or more times.

8. Modifiers

+: Match 1 or more times.

9. Modifiers

?: Match 0 or 1 times.

Example:

1. xxxxxxx | x11

Any 7-digit number; or

011 , 111, 211, 311, 411, 511, 611, 711, 811 , 911

2. [2-8] xxxxxx | 13xxxxxxxxx

begins with "2","3","4","5","6","7" or "8" , follow by 6 digits;

or "begins with 13 and follows by 9 digits"

3. (13 | 15 | 18)xxxxxxxxx

Means to begin with "13","15" or "18" & follow by 8 digits.

4. [1-357-9]xx

Means to begin with "1","2","3" or "5" or "7","8","9" then follow by 2 digits.

4.8.6 Feature Codec

Feature codec includes device function and call function. Feature codec as follow:

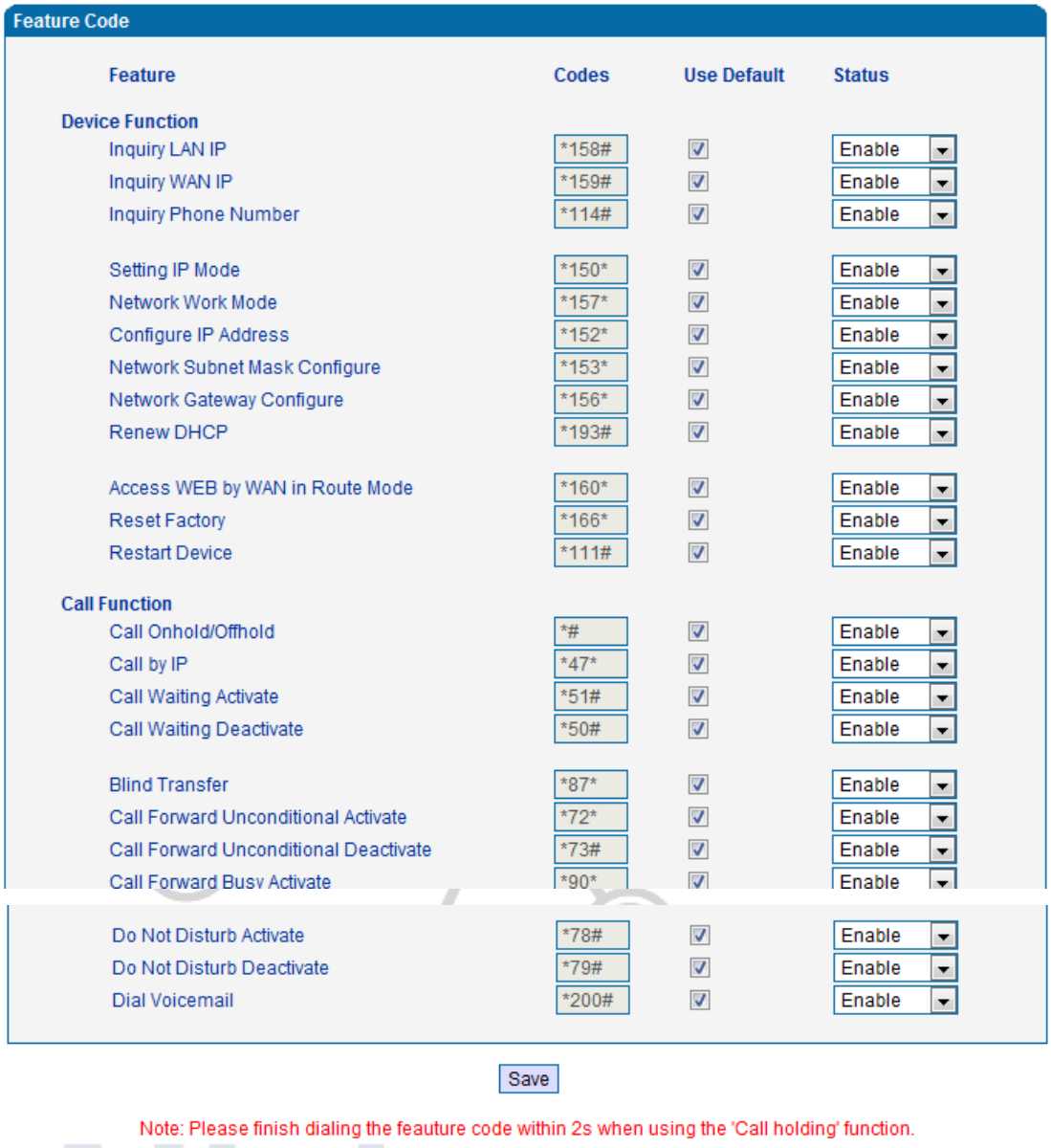

Feature Code
Feature
Codes
Use Default
Status
Device Function
Inquiry LAN IP *158# Enable
Inquiry WAN IP *159# Enable
Inquiry Phone Number *114# Enable
Setting IP Mode *150* Enable
Network Work Mode *157* Enable
Configure IP Address *152* Enable
Network Subnet Mask Configure *153* Enable
Network Gateway Configure *156* Enable
Renew DHCP *193# Enable
Access WEB by WAN in Route Mode *160* Enable
Reset Factory *166* Enable
Restart Device *111# Enable
Call Function
Call Onhold/Offhold *# Enable
Call by IP *47* Enable
Call Waiting Activate *51# Enable
Call Waiting Deactivate *50# Enable
Blind Transfer *87* Enable
Call Forward Unconditional Activate *72* Enable
Call Forward Unconditional Deactivate *73# Enable
Call Forward Busy Activate *90* Enable
Do Not Disturb Activate *78# Enable
Do Not Disturb Deactivate *79# Enable
Dial Voicemail *200# Enable
Save
Note: Please finish dialing the feauture code within 2s when using the 'Call holding' function.

Figure 4.8-10 Feature Code Configuration Interface

Inquire WAN port IP address	Dial*159# to obtain device WAN port IP address
Inquire Phone Number	Dial*114# to obtain port account
Setting IP Mode	*150*0# for ppp modem, *150*1# for static IP, *150*2 for DHCP, *150*3# for pppoe.
Network Work Mode	*157*0#, set network work mode to routing mode; *157*1#, set network work mode to bridge mode
Configure IP Address	*152*+IP, set gateway IP address
Network subnet mask configure	*153*+subnet mask, set gateway subnet mask
Network Gateway Configure	*156*+gateway IP, set gateway

Renew DHCP	*193#, set dynamic IP again
Access Web by Wan in Rout Mode	Allow access web through WAN port：*160*1#; don't allow access web through WAN port：*160*0#
Reset Factory	*166*000000#, reset factory
Restart Device	*111#, restart device
Call onhold/offhold	When call process, dial*# into call hold.（Recovery the call through hook flash or *#）
Call by IP	Directly dial the end user IP to call
Call Waiting Activate	*51#, enable call waiting function
Call Waiting Deactivate	*50#, forbid call waiting function
Blind Transfer	If the call transfer to 801, first hook flash and then dial the * 87 * 801#
Call Forward Unconditional Activate	*72*+ phone number#, transfer the call from the phone number
Call Forward Unconditional Deactivate	*73#, forbid call forward unconditional
Call Forward Busy Activate	*90*+ forward busy number#
Call Forward Busy Deactivate	*91#, forbid call forward busy
Call Forward No Reply Activate	*92*+ forward no reply number#
Call Forward No Reply Deactivate	*93#, close this function
Do Not Disturb Activate	*78#, enable DND function
Do Not Disturb Deactivate	*79#, close DND function
Dial Voicemail	*200#, visit voice mail box

Note：* Private services are open by default

Nicherons

4.8.7 System Parameter

System parameters include: STUN、NTP、Provision、WEB parameter、Telnet.

1）**STUN** (**Session Traversal Utilities for NAT**) is a standardized set of methods and a network protocol to allow an end host to discover its public IP address if it is located behind a NAT. The STUN protocol allows applications operating behind a network address translator (NAT) to discover the presence of the network address translator and to obtain the mapped (public) ip address (NAT address) and port number that the NAT has allocated for the application's UDP connections to remote hosts.

STUN doesn't support TCP connection and H.323.

2）NTP：Network Time Protocol（NTP）is a computer time synchronization protocol.

System parameter configuration interface as follow:

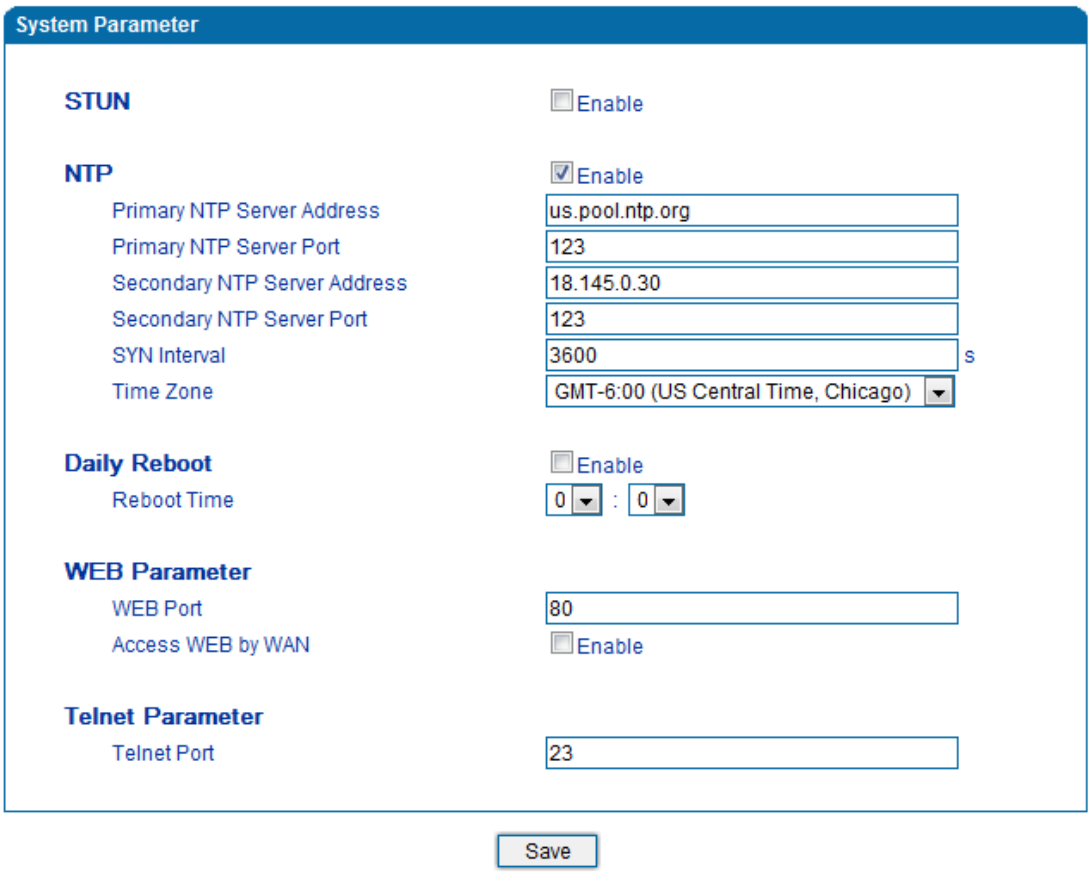


Figure 4.8-11System Configuration Interface

STUN Server Address	STUN server IP address
STUN Server Port	STUN server port
Primary NTP server address	Primary NTP server IP address, system default is us.pool.ntp.org
Primary NTP server port	Default is 123
Secondary NTP server address	Default is 18.145.0.30
Secondary NTP server port	Default is 123
SYN Interval	Every certain time synchronization gateway time, the system default every 3600 s synchronous once.
Time Zone	Time zone can be chosen. System default the United States central

	time, Chicago.
Reboot time	Set a restart time for device, the device will reboot at this time.
WEB Port	Gateway web port, default is 80
Access Web by WAN	Enable or disable accessing web by WAN
Telnet Port	Telnet service port, default is 23.

4.9 Call & Routing

4.9.1 Port Group

Port group parameters include: Index, description etc. Port group configure interface as follow:

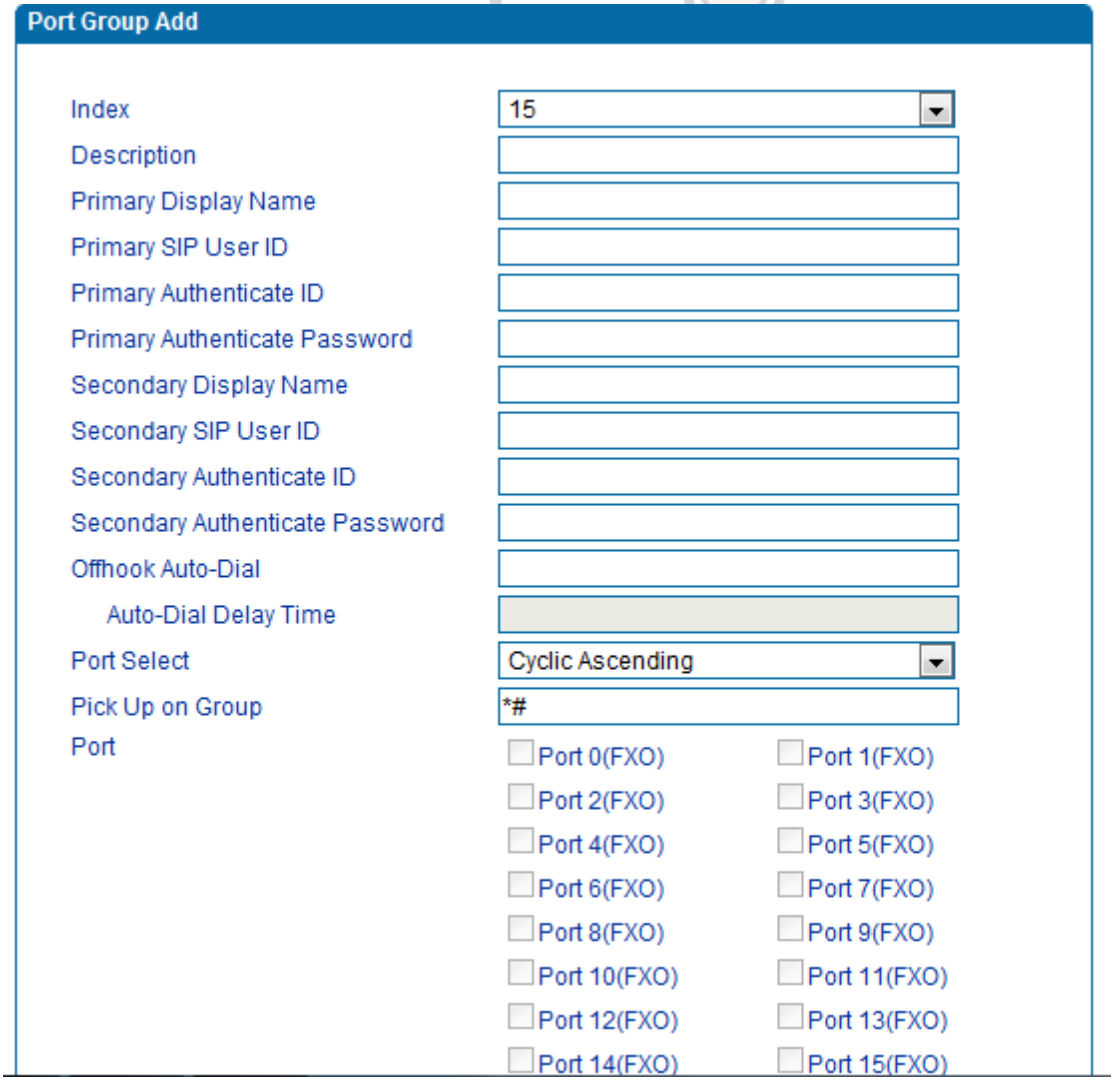


Figure 4.9-1 port group configuration interface

Index	Port groupNumber, It uniquely identifies a route,range

	from 0-15
Description	Port group description,its purpose is so you can identify the port group with a meaningful name
Primary/Secondary Display Name	Port group display, which will be used in SIP message, example: INVITE sip:bob@biloxi.com SIP/2.0 Via:SIP/2.0/UDPpc33.atlanta.com;branch=z9hG4bK776asdhds Max-Forwards: 70 To: Bob <sip:bob@biloxi.com> From: Alice <sip:alice@atlanta.com>;tag=1928301774 Here Bob and Alice is the display
Primary/Secondary SIP User ID	User account information, provided by VoIP service provider (ITSP). Usually in the form of digit similar to phone number or actually a phone number.
Primary/Secondary Authenticate ID	SIP service subscriber's Authenticate ID used for authentication. Can be identical to or different from SIP User ID.
Primary/Secondary Authenticate Password	Password of SIP user ID
Off-hook Auto-Dial	Set Auto-dial number to complete one stage dialing.
Auto-Dial delay time	Delay time of FXO port send auto-dial number.
Port Select	• It specifies the policy for selecting port in a port group • Ascending: the system always selects a port from the minimum number. The preferential selection of the port can be realized through this mode • Cyclic ascending: when system selects ports' Priority, it always begin from the number next to the number selected last time, if the maximum priority number is selected last time, then the next number is the minimum priority number, and move in cycles like this • Descending: when system selects ports' priority, it always begin to select from the maximum priority number • Cyclic descending: when system selects ports' Priority, it always begin from the number before to the number selected last time, if the minimum priority number is selected last time, then the next number is the maximum priority number, and move in cycles like this • Group ring: all ports ringing at the same time
Pick Up on Group	Press "*# +extension number" to decide which extension on the phone.
Port	Add some ports to the same group

4.9.2 IP Trunk

A peer-to-peer VoIP call occurs when two VoIP phones communicate directly over IP without call switch between them. A peer-to-peer call can be initiated directly by dialing destination phone number and also receiving incoming calls from other peer to peer gateway. IP trunk configuration here is to establish peer-to-peer call between SpoATA and other VoIP phones. IP trunk will be used in routing configuration.

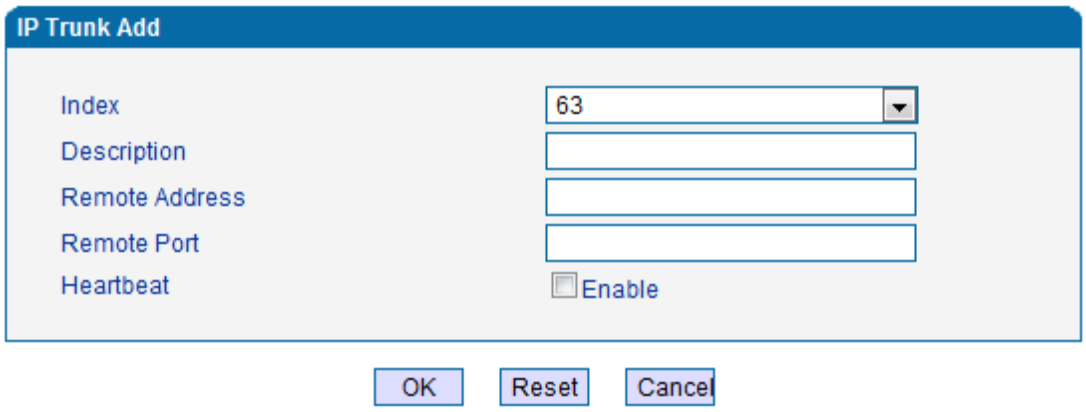
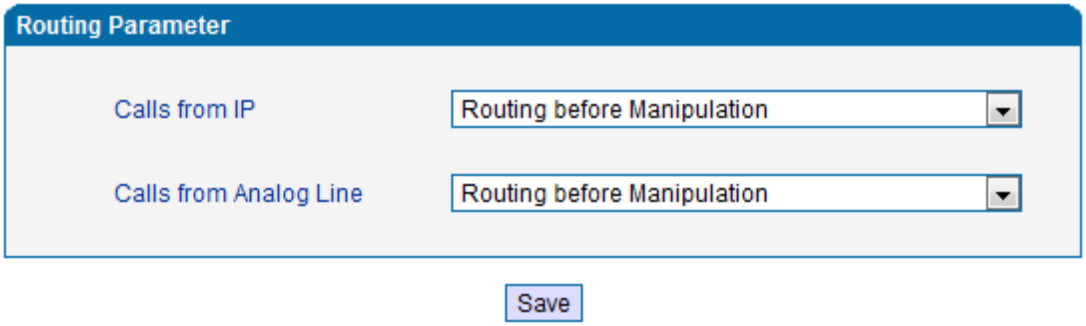


Figure 4.9-2 IP Trunk Configuration Interface

Index	IP trunk number, it is range from 0 to 63
Description	The description of IP trunk, its purpose is so you can identify the IP trunk with a meaningful name
Remote Address	Peer IP address or domain name
Remote Port	Peer SIP port
Heartbeat	Default is disable, if enable, ATA will send "OPTION" to peer device

4.9.3 Routing Configuration

Figure 4.9-3 Routing Parameter Configuration Interface



This option determines the following routing of call take effect before or after manipulation.

4.9.4 IP-Tel Routing

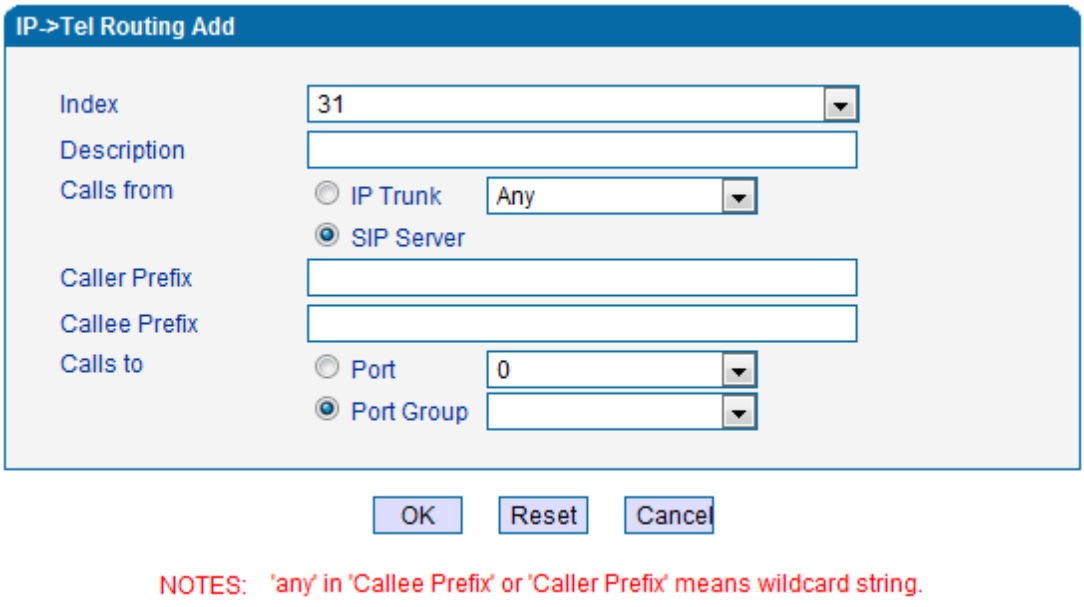


Figure 4.9-4 IP-Tel Routing Parameter

Index	Routing priority: 0-31, 0 is the highest priority.
Description	its purpose is so you can identify theIP0->Tel routing with a meaningful name
Calls from	IP Trunk/SIP Server, any means any IP
Caller Prefix	Caller number Prefix, its length normally less or equal to caller number, which helps to matching routing exactly. if caller number is 2001, the caller prefix can be 200 or 2. “any” means match any caller number like “bob1”,”29801”
Caller Prefix	Called number Prefix, its length normally less or equal to called number, which helps to matching routing exactly. if called number is 0019052050512, the called prefix can be 001905 or 00.,“any” means match any called number
Calls to	This call routing is routing to port or port group

4.9.5 Tel-IP/Tel Routing

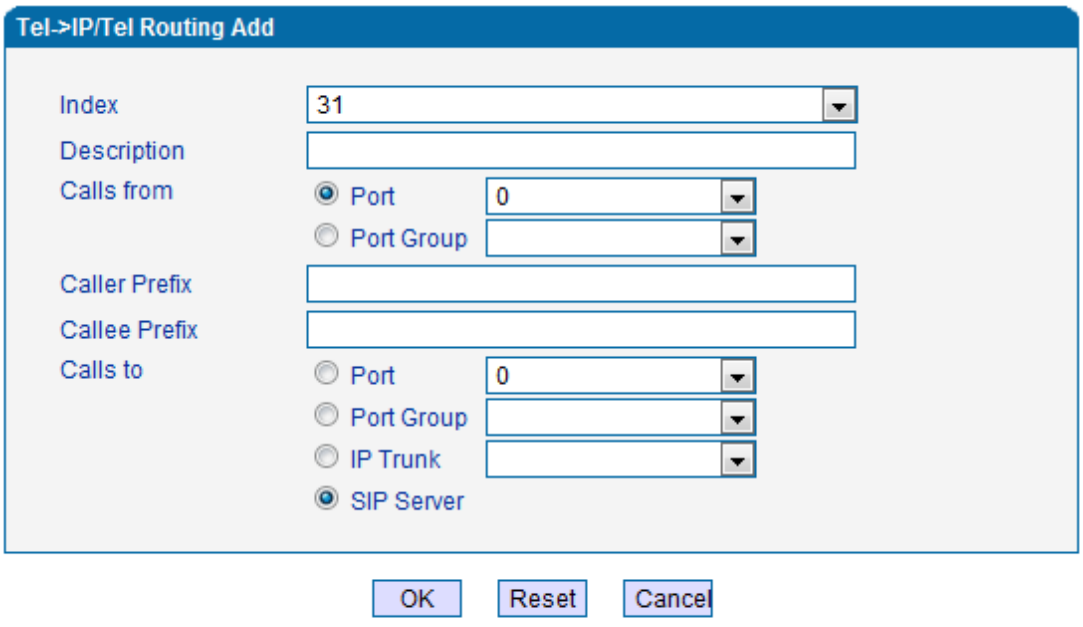


Figure 4.9-5 Tel-IP/Tel Parameters Configuration

Index	Routing priority :0-31, 0 is the highest priority.
Description	its purpose is so you can identify the routing with a meaningful name
Calls From	Tel-IP call select port or port group
Caller Prefix	Caller number Prefix, its length normally less or equal to caller number, which helps to matching routing exactly. if caller number is 2001, the caller prefix can be 200 or 2. “any” means match any caller number like “bob1”,”29801”
Caller Prefix	Called number Prefix, its length normally less or equal to called number, which helps to matching routing exactly. if called number is 0019052050512, the called prefix can be 001905 or 00., “any” means match any called number
Calls to	This call routing is routing to port, port group, IP trunk and SIP server.

4.10 Manipulation Configuration

4.10.1 IP-Tel Caller

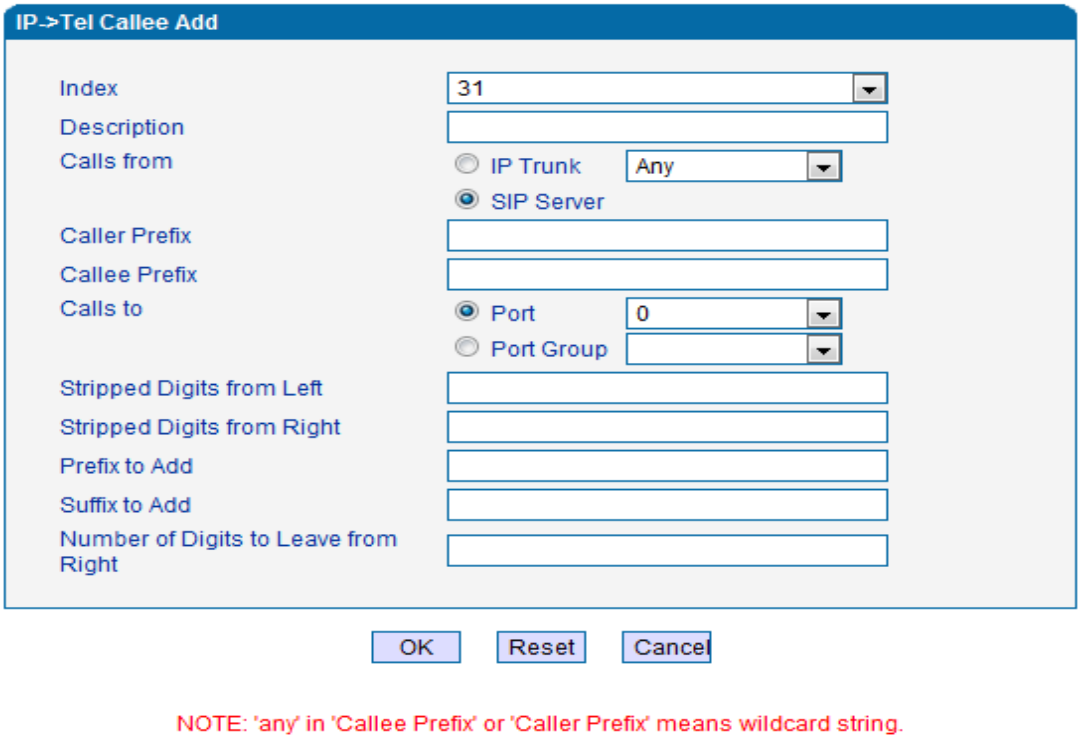


Figure 4.10-1 IP-Tel Caller number configuration

Calls From	This call come from IP trunk or SIP server.
Caller Prefix	Caller number Prefix, its length normally less or equal to caller number, which helps to matching routing exactly. if caller number is 2001, the caller prefix can be 200 or 2. "any" means match any caller number like "bob1","29801"
Caller Prefix	Called number Prefix, its length normally less or equal to called number, which helps to matching routing exactly. if called number is 0019052050512, the called prefix can be 001905 or 00., "any" means match any called number
Calls to	This call routing is routing to port, port group
Stripped Digits from Left	Remove the called number digits from the left
Stripped Digits from Right	Remove the called number digits from the right
Prefix to Add	Add a number prefix
Suffix to Add	Add a number suffix

Number of Digits to Leave from Right	Starting from the right to retain the called number digits

4.10.2 Tel-IP Caller

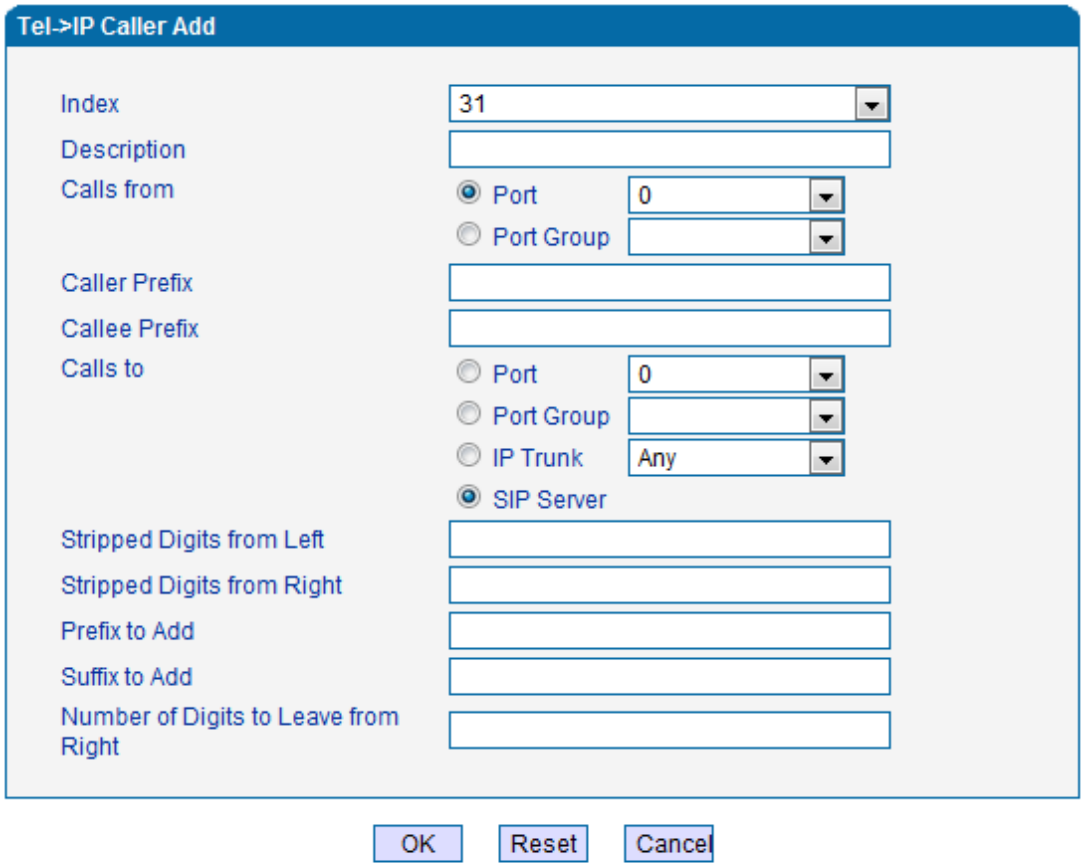


Figure 4. 10-2 Tel-IP Caller

Configuration parameters are the same with "IP->Tel Caller".

4.10.3 Tel-IP Caller

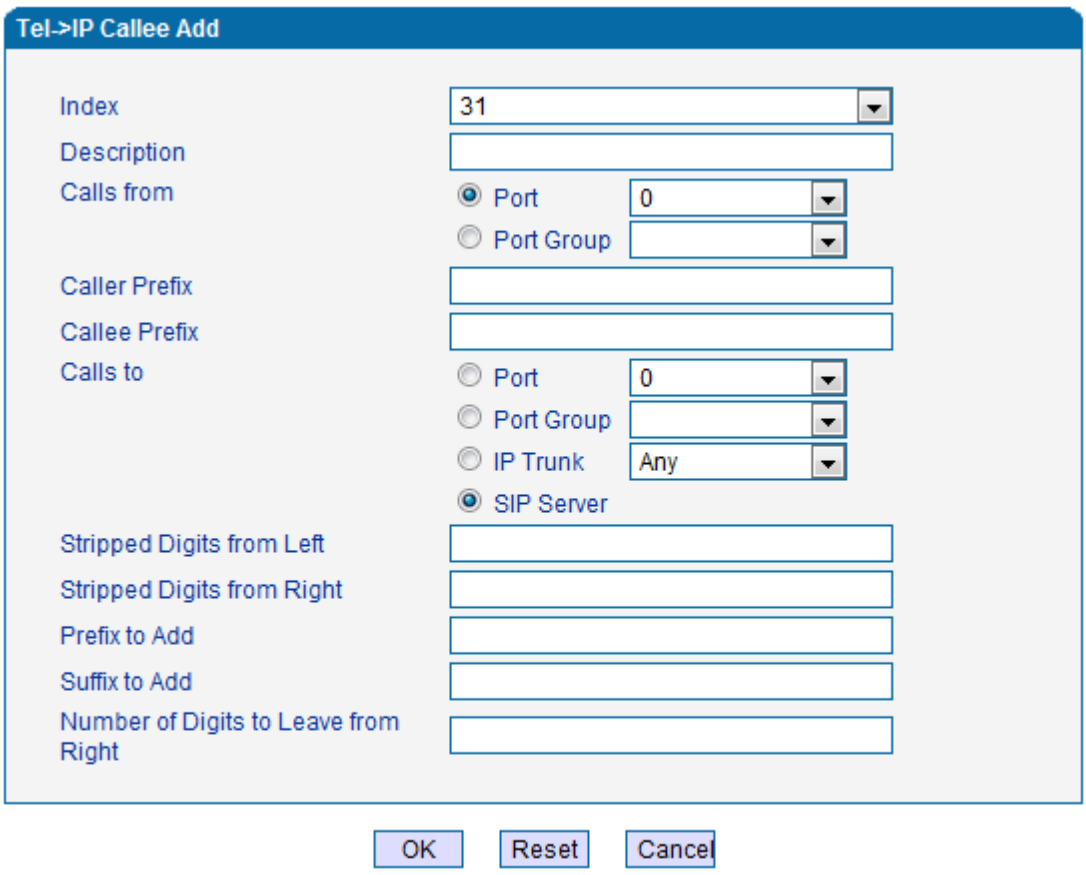


Figure 4.10-3 Tel-IP Caller

Configuration parameters are the same with "Tel->IP Caller".

4.11 Maintenance

4.11.1 syslog Parameter

Syslog is a client/server protocol: a logging application transmits a maximum 1024-byte text message to the syslog receiver. The receiver is commonly called syslogd, syslog daemon or syslog server. Syslog messages may be sent via the User Datagram Protocol (UDP) or the Transmission Control Protocol (TCP).

Syslog level：

- none Used to mis-arrange

- debug Not including function conditions or the question of other information
- notice importance common conditions
- warning Early warning information
- error Stop error conditions of tools or some part of the realization of the function subsystem

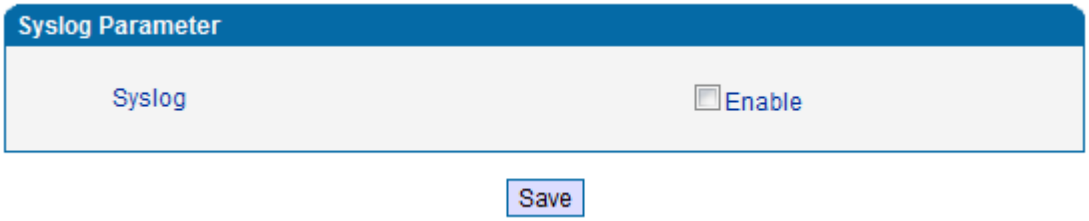


Figure 4.11-1 Syslog Parameter Configuration

Enable send CDR, and then send communication information to syslog server.

4.11.2 Firmware Upload

The process of firmware upload：

1）Click "Firmware Upload"

2) Browse files and choose the loading program (Name the file extension. ldf)

3) Click "Upload", the upload process will last about 60s and device will automatically restart after uploading.（During the firmware update process, please don't shut down the power）

4.11.3 Data Backup

The process data backup:

1) Click "Data Backup"

2) Click "Backup" to backup data to PC.

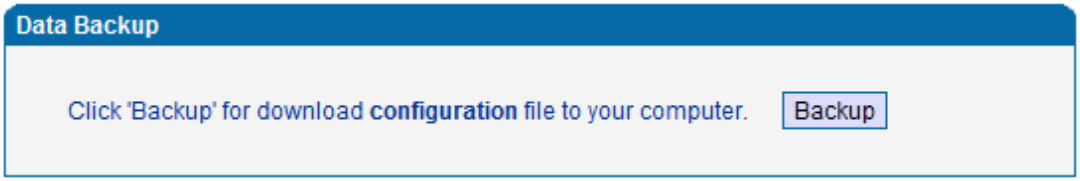


Figure 4.11-3 Data Backup Interface

4.11.4 Data Restore

The processes of data restore:

1) Click "Data Restore"
2) Browse file, select data file.
3) Click "Restore" and then import successfully, the device will restart automatically.

Figure 4.11-4 Data Restore Interface

4.11.5 Ping Test

Ping is a computer network administration utility used to test the reachability of a host on an Internet Protocol (IP) network and to measure the round-trip time for messages sent from the originating host to a destination computer. Ping instructions:

1) Click "ping test"
2) Enter IP address or domain connected, click start.
3) Received a message indicates that network connection normal, or network connected to a fault.

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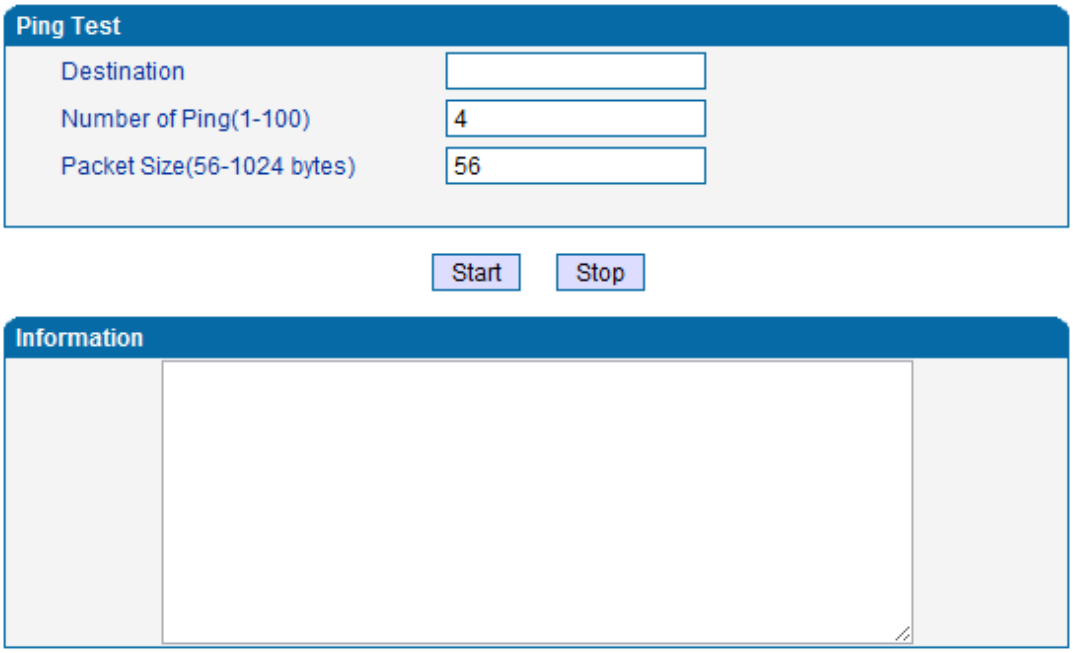

Figure 4.11-5 Ping Parameter Interface

4.11.6 Tracert Test

traceroute is a computer network diagnostic tool for displaying the route (path) and measuring transit delays of packets across an Internet Protocol (IP) network. Traceroute sends a sequence of three Internet Control Message Protocol (ICMP) echo request packets addressed to a destination host. The time-to-live (TTL) value, also known as hop limit, is used in determining the intermediate routers being traversed towards the destination.

Tracert introduce:

1) Click tracert test.
2) Fill IP address or domain connected, click start.

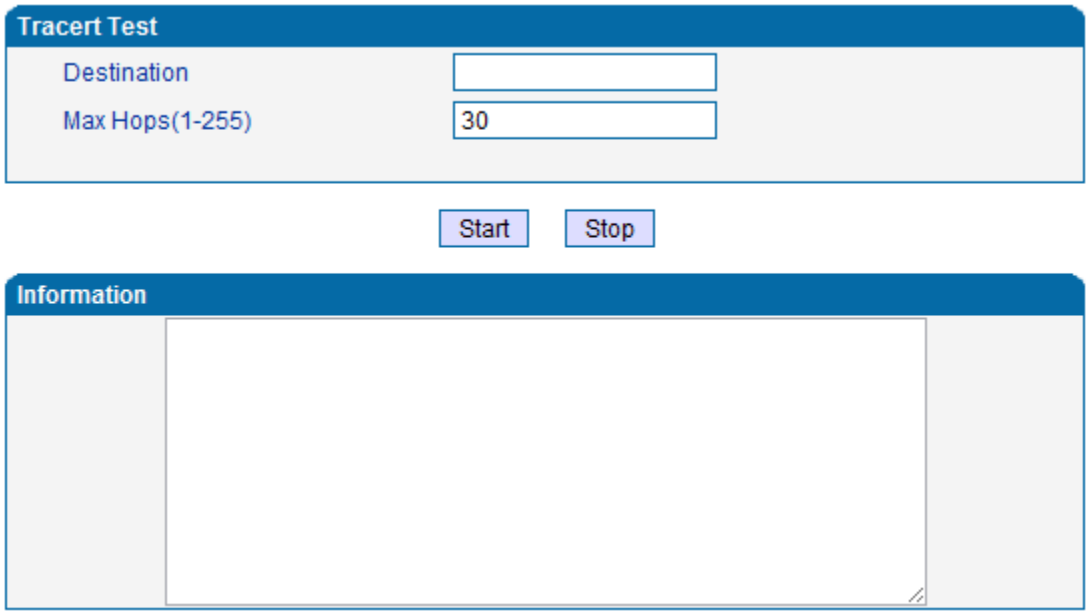

Figure 4.11-6 Tracert Test Interface

4.11.7 Password Modification

Includes WEB username and password, Telenet username and password modify.

Note：Default web and telnet username and password is: admin, admin.

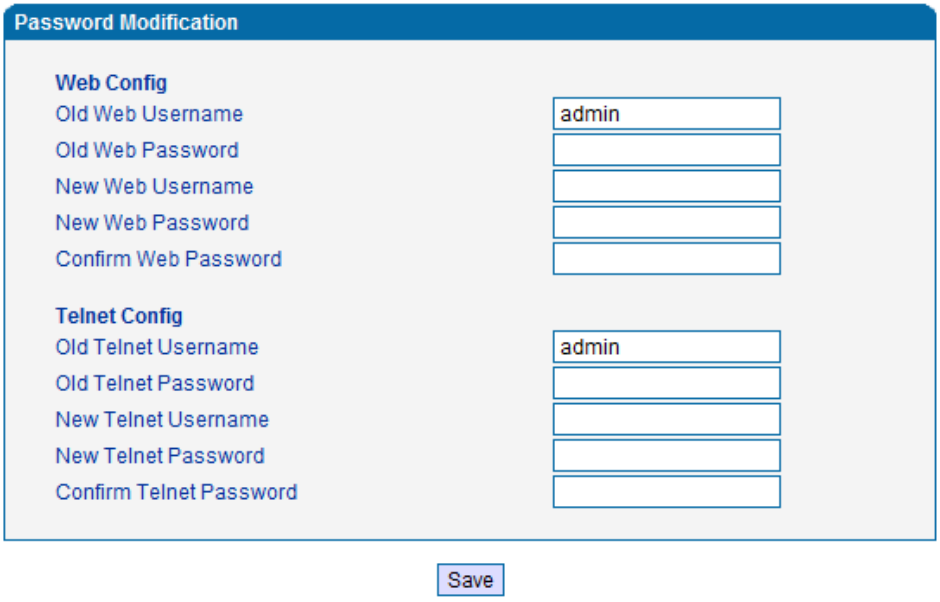

Figure 4.11-7 Password Modification Interface

4.11.8 Factory Reset

Click "Apply" to restore the factory settings.

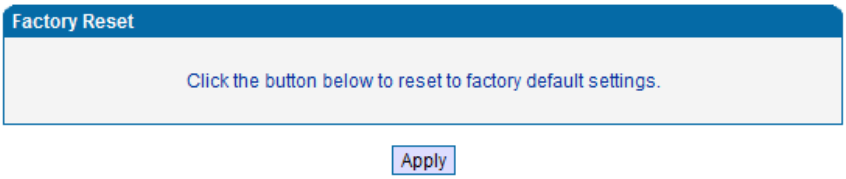

Figure 4. 11-8 Factory Reset Interface

4.11.9 Device Restart

Certain configuration changes will require a reboot before applying to the gateway.

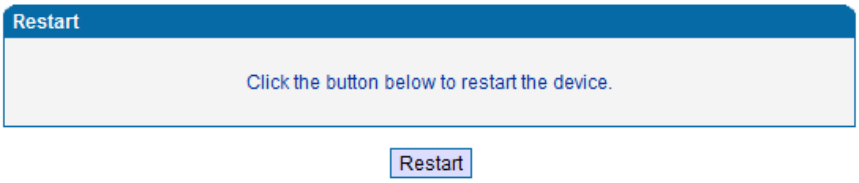

Figure 4.11-9 Device Restart

5. Glossary

- DNS：Domain Name System
- SIP：Session Initiation Protocol
- TCP：Transmission Control Protocol
- UDP：User Datagram Protocol
- RTP：Real Time Protocol
- PPPOE：point-to-point protocol over Ethernet
- VLAN：Virtual Local Area Network
- ARP：AddressResolution Protocol
- CID：Caller Identity
- DND：Do NOT Disturb
- DTMF：Dual Tone Multi Frequency
- NTP：Network Time Protocol
- DMZ：Demilitarized Zone
- STUN：Simple Traversal of UDP over NAT
- PSTN：Public Switched Telephone Network

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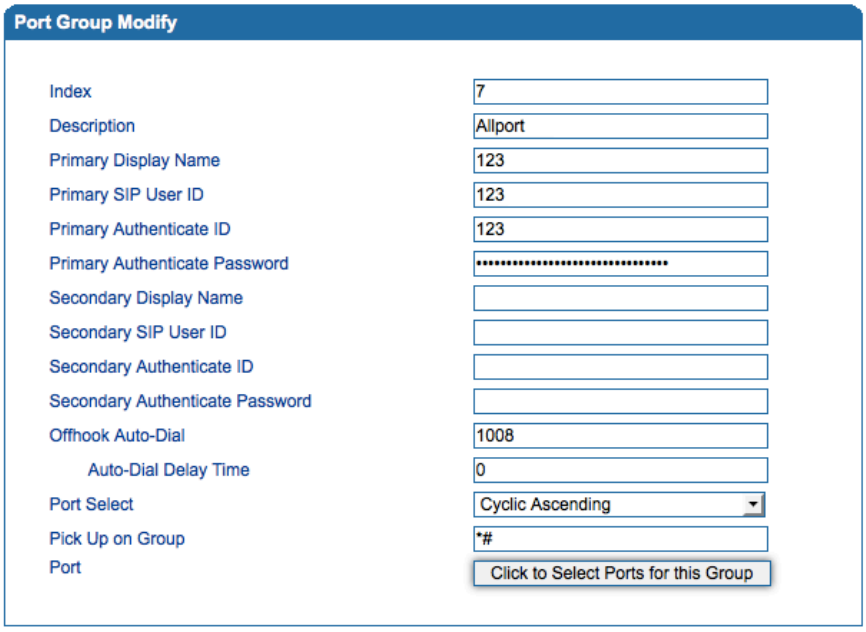
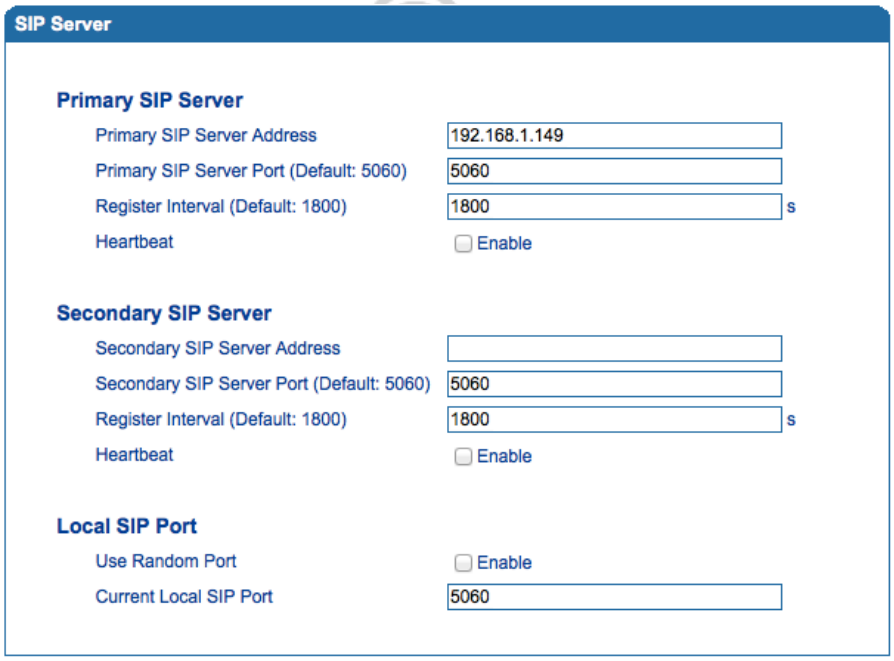
6. Configuration Examples

1) Elastix/Trixbox/Freepbx:
PBX ip: 192.168.1.149 ; Username= 123 ; password=123,
Ringgroup@Elastix=1008

Call & Routing-→ Port Group



SIP server→



Call & Routing-→ IP Trunk

IP Trunk

	Index	Description	Remote Address	Remote Port	Heartbeat
☐	63	elastix	192.168.1.149	5060	Disable

Call & Routing-→ "IP ->Tel Routing"

IP->Tel Routing

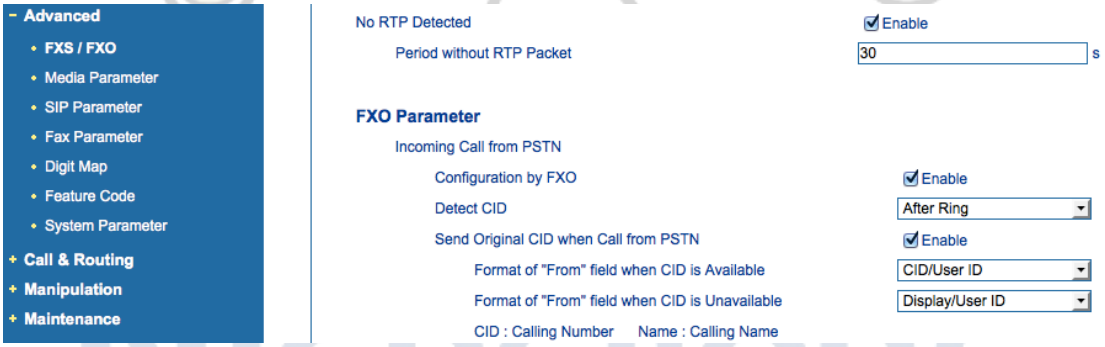
	Index	Description	Calls from	Caller Prefix	Callee Prefix	Calls to
☐	31	allport	Any	any	any	Port Group 7

Call & Routing-→ "Tel ->iP Routing"

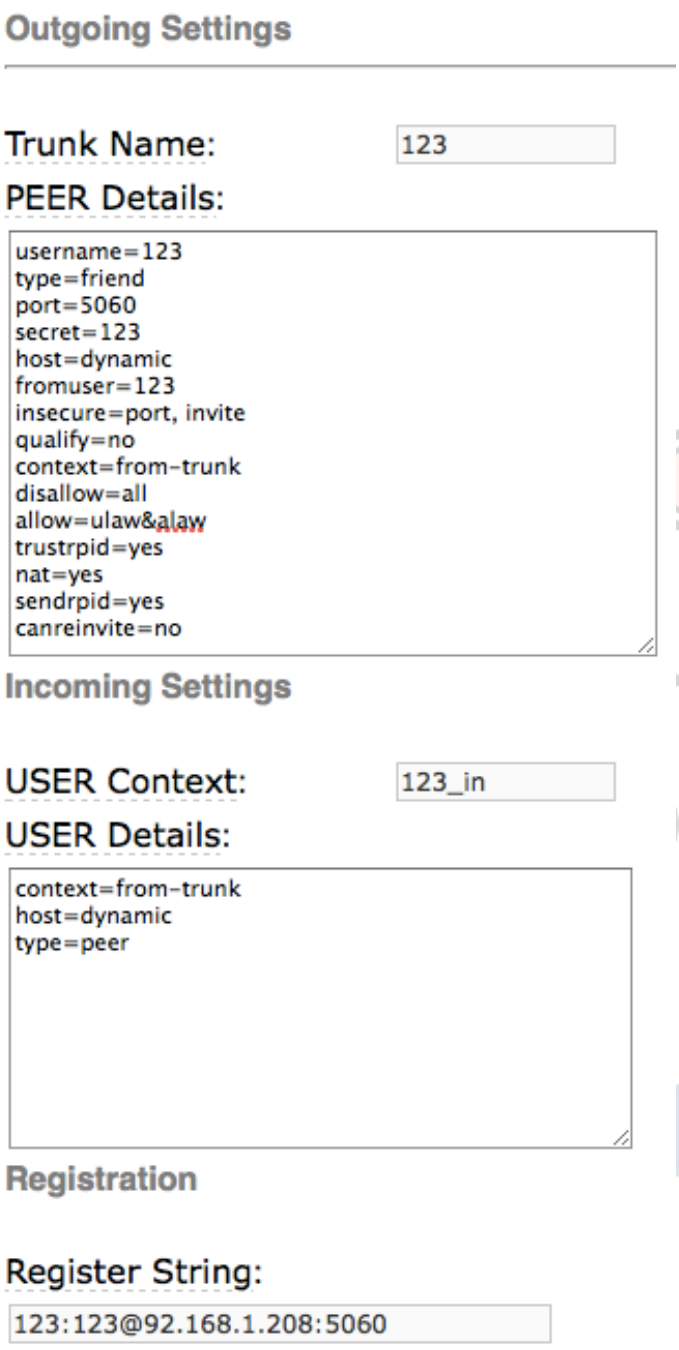
Tel->IP/Tel Routing

	Index	Description	Calls from	Caller Prefix	Callee Prefix	Calls to
☐	31	PBX	Any	any	any	IP Trunk 63 <elastix>

Advanced→FXS/FXO→ FXO Parameter



Elastix→ Trunk→ SIP trunk

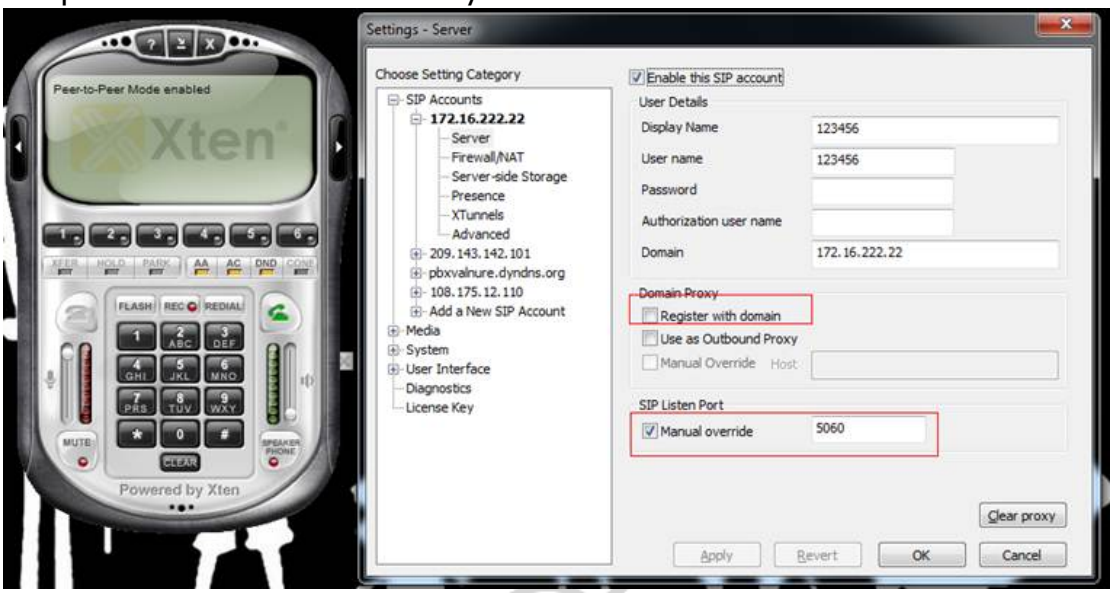


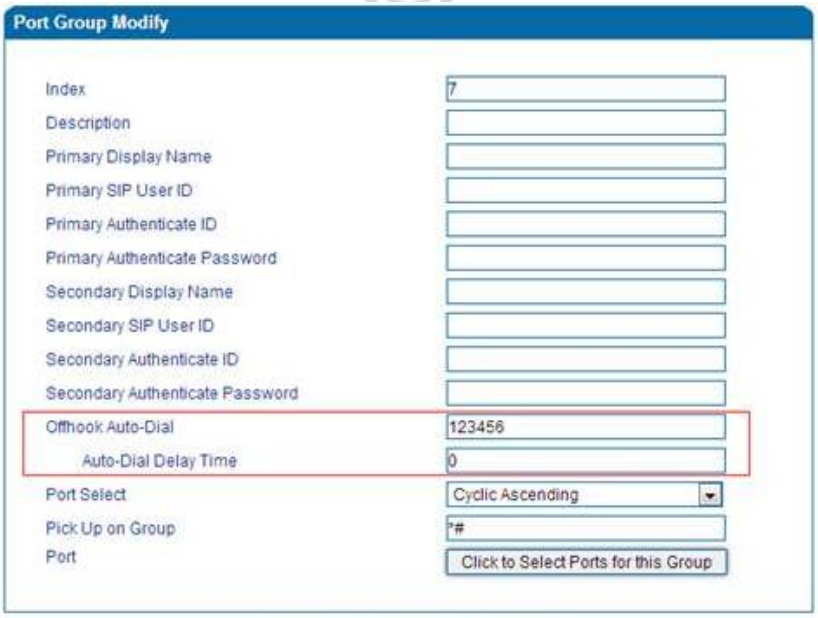
Gateway Status & Statistics→ Registration

Port Group Registration Information

Port Group	Port	Primary User ID	Primary User Status
7 <portgroup>	0,1,2,3,4,5,6,7,	123	Registered

1) Softphone or other ATA:
SIP port must be set to 5060 only





Tel->IP/Tel Routing

	Index	Description	Calls from	Caller Prefix	Callee Prefix	Calls to
☐	31	Tel->IP	Any	any	any	IP Trunk 63 <ToSoftphone>

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